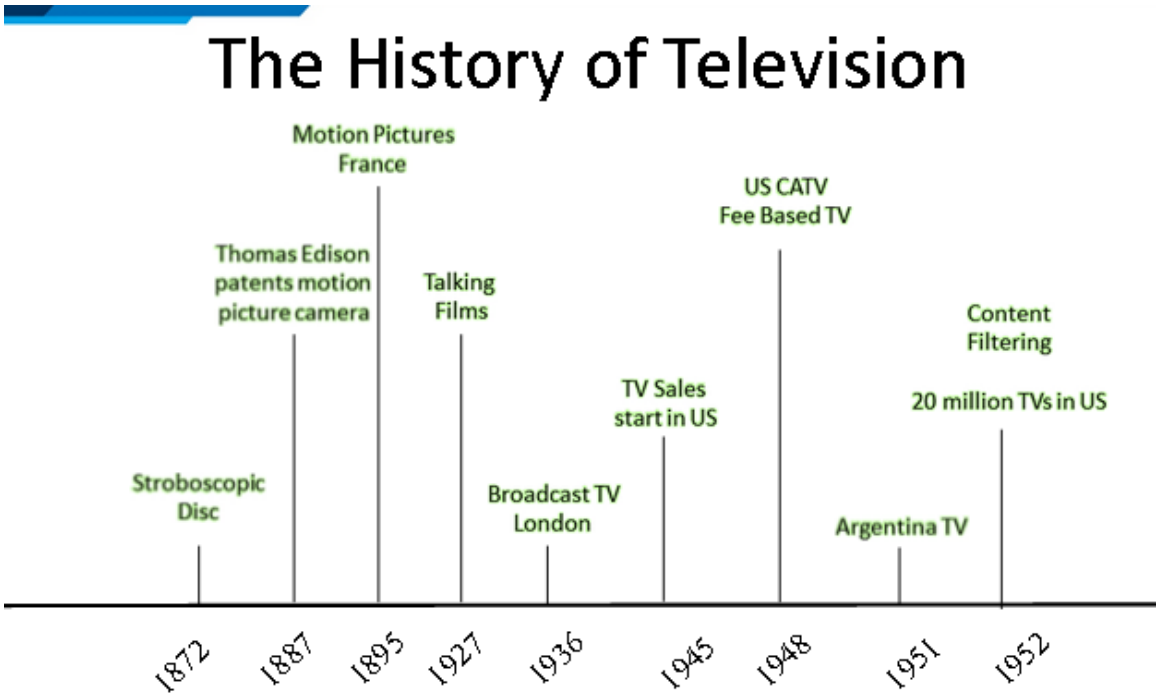
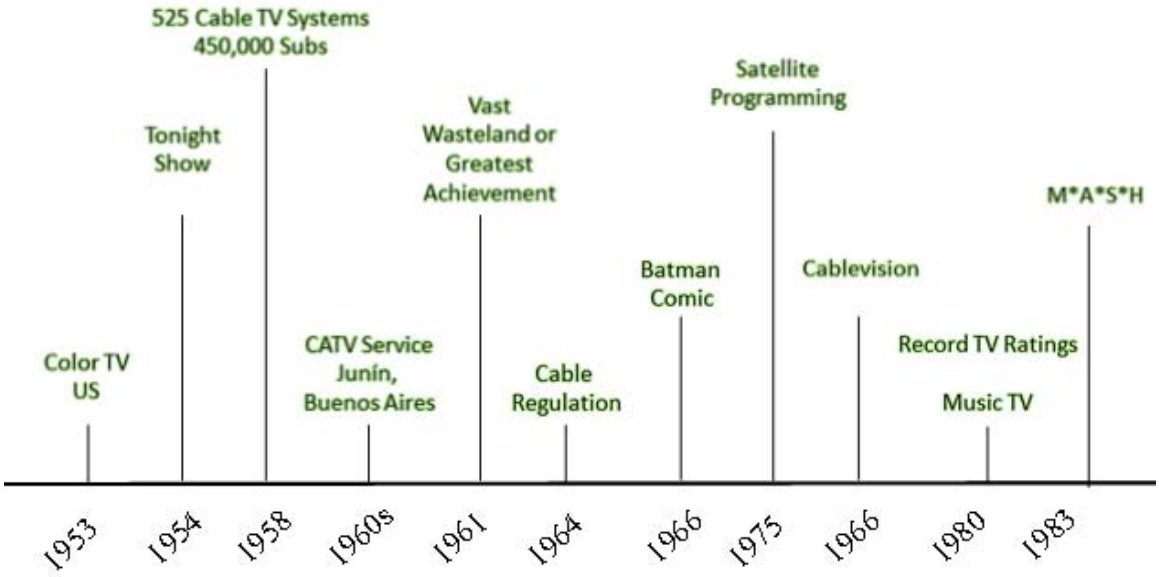


let's explore a brief history of television as we can see here in 1872 and started a series of photographs called Stroboscopic so its were TV a guess in a sense started as you see in 1887 Thomas Edison has a couple pads here for motion picture camera the 1st public demonstration of motion pictures was 1895 1927 talking films I started to begin with L Jolson 1936 1st television broadcast made available in London, 1945 gimbal department store was ready to sell 1st large-scale televisions 25,000 people came over 3 weeks for it she has to watch NBC programs from New York sent out by the Philco Philadelphia station in 1948 the Ed Sullivan show appeared and variety of shows continue to air on channels like CBS well into 1971 spurring advancements in the 2 different shows in and show business careers in 1948 the earliest cable systems were born areas of Pennsylvania and Oregon of the United States in 1934 was early communication act for for not broadcasters, where the FCC adopts its fairness doctrine in 1949 a 1951 or just tell the Argentina and South America began its broadcasting and 1952 in response to protests some people thought TV was a immoral is very interesting and by the end of 1952 there was over 20 million TV households so that was quite quite some growth from 1945



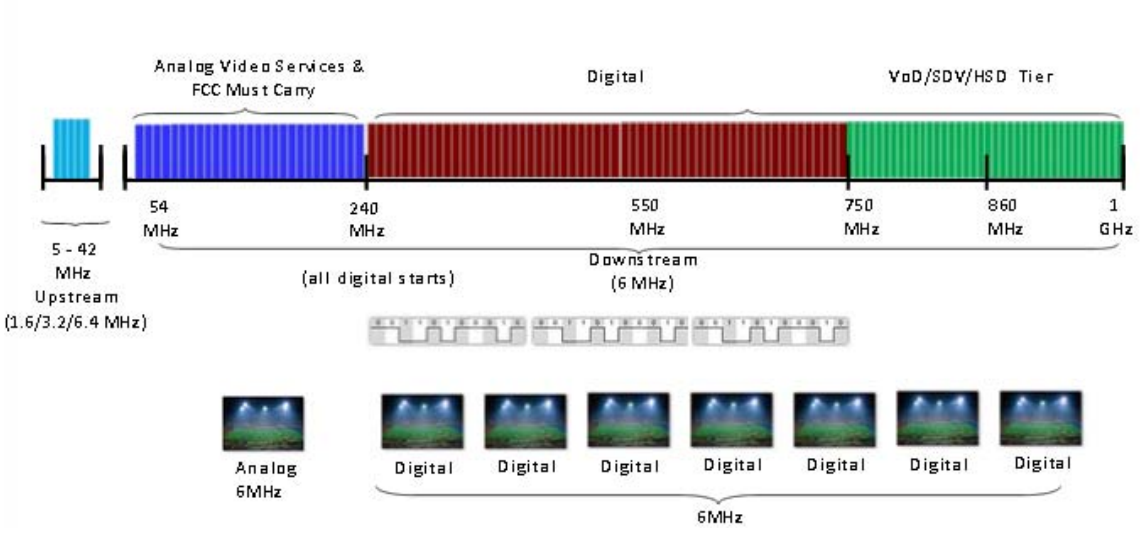
1953 color broadcasting arrives so this is a good milestone in the history of television of 1954 NBC launches the Tonight Show so we have the 1st time a comedian will be on television in 1958 there are 525 cable TV system serving 450,000 subscribers as of things are certain pickup in the cable industry in the 60s cable TV started in South America about 5.5 million homes in Latin America and buenas arasArgentina is the 2nd largest market in the world 1960 and 51 we can see that there are some more controversy around television there is a speech given by Newton me know if it says that the TVs are vast wasteland but then the vice president.Herbert Humphrey: TV one of the greatest single achievements in communication are 64 FCC issues him cable regulation 66 live action represented on a comic strip Batman appears on television 75 time initiates the concept of linking satellite programming to cable systems so that we can launch services like Home Box Office and see heavyweight boxing championships like Joe Frazier and Mohammed ali broadcast from Manila and 79 Cablevision starts in South America cable operator and many cable operators are starting a cross like for instance Comcast has its roots in from 1948 1980 who shot JR another big hit TV show 1980 MTV comes on the scene 83 final episode of M*A*S*H 125 million homes tuned in to watch this 30 minute show which had a final finale of 2 1/2 hours 89 paper view becomes familiar part of television service in 1992 rebuild of cable systems begin to form what we call today is the HFC increasing the ban with the 700 MHz 94 field testing of HDTV and then we see 95 the FCC officially sets the standard for HDTV 96 digital satellite dishes are now only 18 inches in diameter at the market to become the biggest selling electronic item in history next to VCRs in 98 cable systems begin widescale testing and deployment of digital cable to compete with satellite television in 2008 some movement down in South America as far AS cable operators consuming other cable operators 2009 TV stations across the US sees transmission of a log over the year and 2010 we see some some 3-D television happening in 2012 10 2013 were talking about a 4K television

cable operators use digital signals to increase the quality and quantity of subscriber services offered at sig look

cable television started as a way for communities isolated from over the air broadcasts by

geographic features or distance to receive television video signals the system was referred to as community antenna television or CATV cable systems have since expanded services beyond over the year today large cable operators called multisystem operators have expanded their cable networks across multiple states a trend in cable networks to interconnect the many headends hubs and optical transfer nodes with a fiber transport network to share digital video content providers using Internet protocol and moving pictures expert group are able to transport video signals over the fiber-optic transport the use of IP MPEG and optical provides a reduction in the equipment at each headend to maintain video services it also simplifies delivery or delivering constant quality of service for all cable customers

there are many types of digital video content available to the cable subscriber today whether it's HD broadcast digital video SD broadcast digital digital video which would be high definition and standard definition or narrowcast services like video-on-demand we also have paper view enhance television interactive television and the media that you would get on this weathers DVD or CD-ROM we also have other types of video emerging called over-the-top the rise over top of an IP network and that we have Internet protocol television which is more of a managed video service and switched digital video



in the mid-to-late 1990s cable operator began to deploy digital tier of cable the area in maroon shown here on the screen digital was necessary to compete and to free up precious RF spectrum space in the HFC or outside plant today operators can converge all the subscriber services digital voice digital data digital video together on the same digital network digital offers speed of manipulation and storage retrieval for virtual error-free recreation from the original content for the operator in the analog world only one analog video signal could be sent in a 6 MHz channel and you can see this across the bottom one single channel for 6 MHz but when the video signal is digitized and compressed with MPEG 8 standard 2 high definitions or 2 3-D or one older definition signal may be possible meaning that each 6 MHz analog can be converted to 8 standard definition or to HD or 3-D 2 of those were one ultra so here we see a single 6 MHz

analog can be converted to 7 digital video signals

digital is defined as an on or and off also known as 01 called binary the stream of meaningful zeros and ones can be sent to a destination device or in our case a digital set-top box with a destination device is able to read the language of zeros and ones and do what is requested in the transmission language used by cable operators is commonly known as MPEG MPEG allows analog video and audio to be digitized sent as a transport stream using MPEG in the world of analog video something more than a collection of still pictures called frames when the frames are presented in quick succession on a video displayed appears to be fluid motion videos typically captured at 29.97 frames per 2nd the world of digital works much the same way as analog streams a stream of ones and zeros called bits of data are sent to a decoder in our case the set-top box and this will reassemble those binary digits to create the 29.97 frames per 2nd the difference in the video is that the data is in bits the smallest addressable element of the video display in our case a television is called a pixel and must be represented by these digital bits digitization or encoding is the conversion of analog data into digital data digital data is a sequence of discrete disc discontinuous voltage pulses digital data takes less space on the age of the spectrum to transport and less space on servers to store

a coder is a device used to convert analog information into digital information so here we see the analog information be encoded into zeros and ones and then decoded on your side a decoder is a device used to convert digital information back to analog information a codec performs both encoding and decoding

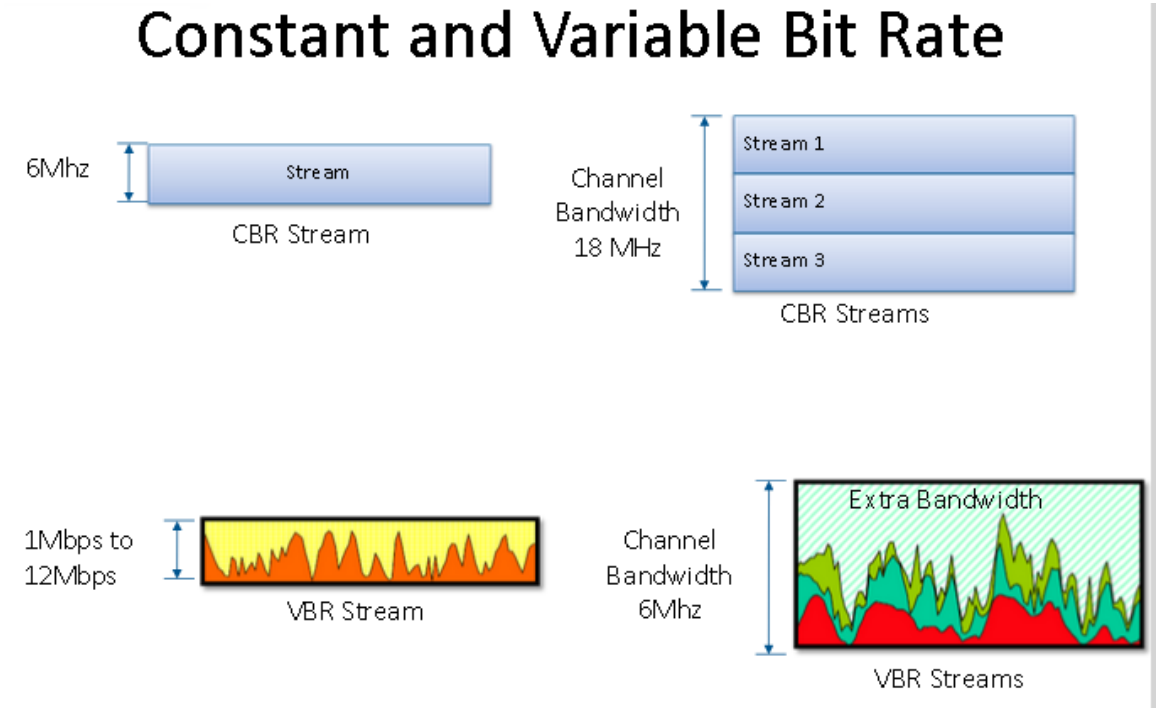
pixels are individual luminescent dots that were displayed correctly create images brightness intensity and color information is encoded into a string of data for each pixel

digital channels are still 6 MHz in the United States and maybe 8 MHz wide outside United States 6 MHz wide channel may contain 6 to 8 digital services 6 megahertz digital channels are sometimes referred to as QAM

in interlaced scanning and images produced by illuminating individual horizontal lines beginning with lines 1 3 5 etc. and then illuminating lines 2 4 6 etc. the process is from left to right starting at the top and working towards the bottom of the screen shown here standard analog television sets utilize 480 visible horizontal lines to create a single frame of video high-definition digital television or HDTV these sets may utilize 1080 interlaced horizontal lines or 1080i now many programmers may use attending the I format however content is moving in the direction of 1080 P and 2160 P

another TV video format is progressive scanning progressive scanning produces a frame of pixels sequentially for example lines 1 2 3 4 etc. working left or right and from top to bottom the 720 P format is one of example of this format in this format 720 lines are produced sequentially to create a full frame some programmers use the 720 P format it should be noted that some television sets have a preferred format for instance of the preferred format is 720 P things that will convert any received 1080 I signal to the 720 P format besides a 720 P progressive scanning format 1080 P 480 P and 2160 P format exists the 480 P is not an HD format it was used for DVD video Blu-ray is example of the 1080 P while ultra high definition or you UHD is an example of 2160 P in a 6 MHz channel 2 to 3 720 P video streams may coexist 2 to 310 80 P video streams

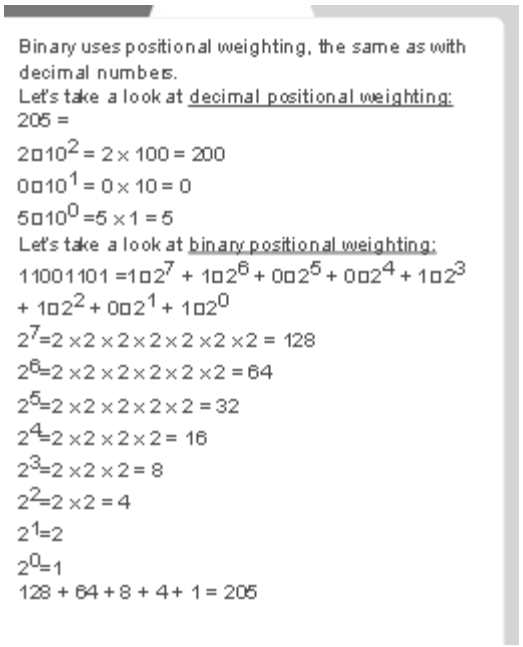
may be possible and 2 2160 P video streams may exist



there are 2 types of bit rates used by cable operators constant bit rate and variable bit rate in constant bit rate the rate of the transport stream consumption is contained in the decoder the constant bit rate usually is created by stuffing null MPEG frames into the transport stream to constant is always the same notices on the screen here we have a CBR 6 MHz when combined with other CBR streams it becomes 18 MHz

in variable bit rate the video encoder varies the amount of output data per time segment more bits are allocated to more complex content variable bit rate uses less overall bandwidth as shown here we have a variable bit rate stream that goes anywhere between 1 and 12 but in a single 6 MHz channel we could have several variable bit rate streams

digital signals use the binary system were base 2 a binary code is a way of representing numeric values using 2 symbols or states ones and zeros these digital signals are the same types of signals used for computers and these are the types of signals used for digital video and a cable operators network let's take a look.


Binary uses positional weighting, the same as with decimal numbers.
Let's take a look at decimal positional weighting:
205 =
2□10^2 = 2 × 100 = 200
0□10^1 = 0 × 10 = 0
5□10^0 =5 × 1 = 5
Let's take a look at binary positional weighting:
11001101 =1□2^7 + 1□2^6 + 0□2^5 + 0□2^4 + 1□2^3
+ 1□2^2 + 0□2^1 + 1□2^0
2^7=2 ×2 ×2 ×2 ×2 ×2 ×2 = 128
2^6=2 ×2 ×2 ×2 ×2 ×2 = 64
2^5=2 ×2 ×2 ×2 ×2 = 32
2^4=2 ×2 ×2 ×2 = 16
2^3=2 ×2 ×2 = 8
2^2=2 ×2 = 4
2^1=2
2^0=1
128 + 64 + 8 + 4 + 1 = 205

Decimal positional weighting:

$$205 = 2\times10^2\ (100) + 0\times10^1\ (10) + 5\times10^0(1)$$

$$200 + 0 + 5 = 205$$

decimal positional waiting lets say we have a decimal value of 205 205 if we break it out into the powers of 10 since decimal of 0 through 9 base 10 2×10 to the 2nd since 10 the 2nd is 100 would yield 200 0×10 to the 1st verses of 10 to 0×1005×10 zeros anything to the 0 power is always one to 5×1 is 5 and then we add among 200+0+5 is 205

Binary positional weighting:

$$11001101 =$$

$$1\times2^7 + 1\times2^6 + 0\times2^5 + 0\times2^4 + 1\times2^3 + 1\times2^2 + 0\times2^1 + 1\times2^0$$

lets take a look at binary positional waiting to hear I have a series 8 bits so a bit in binary can either be a 1 or 0 and bit him 8 of citizens we have 8 really talking about a because of octet and octet is is this the different sections of that IP address in each section is worth a bit so stick with how this plays out so we have the 1st binary value is one so that in the 7th position so if you look

always the right of this binary number is the 0 position always the left of binary number would be the 7th 8 positions so sorting from the left one times to the 7th is a one-time through the 7th is really yield a 128 1 times through the 6 yield 64+0 so the 0 times a day 0 was another 00 times anything is one of the 0 then we have a 1 times through the 3rd book to Dr. Stuart is 38 8 and we have one times through the 2nd 3 seconds for we have another 00 diving 0 then there's attitude of the 0 and so that was once one times 1 is when you give us one so how does this all out out what is this me walk by the binary number here were converted to decimal this is the chart to do so or look at a magic box, not

Binary positional weighting:

$11001101 =$

$$1\times2^7 + 1\times2^6 + 0\times2^5 + 0\times2^4 + 1\times2^3 + 1\times2^2 + 0\times2^1 + 1\times2^0$$

$$128 + 64 + 8 + 4 + 1 = 205$$

so here's what we get 2 to the 7th values 128 through the 6 value 64 2 to the 4th value is 8 2 to 3rd value to to the 3rd value is 8 to the 2nd value was 4 and then to the 0 value is 1 and was at that up SR 205

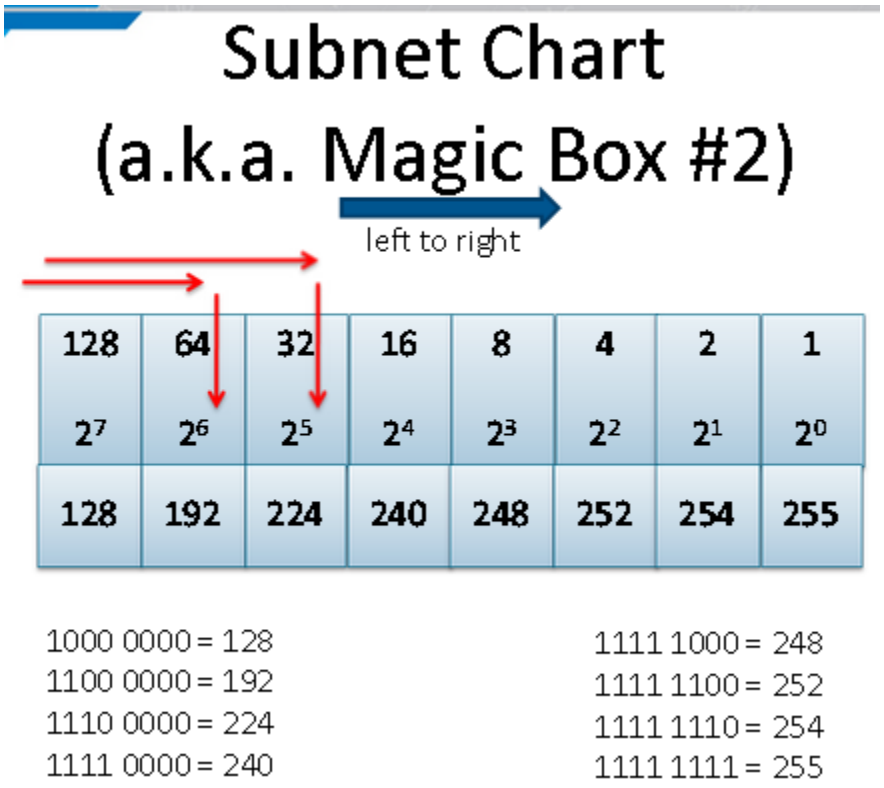
	128	64	32	16	8	4	2	1
	2^7	2^6	2^5	2^4	2^3	2^2	2^1	2^0
192	1	1	0	0	0	0	0	0

Values available from the chart above:

0 to 255 in decimal or 0000 0000 to 1111 1111 in binary

since each octet is 8 bits we can use the following base to chart a.k.a. the magic box to convert decimal to binary and minor decimal this box represents every possible number in a single IP address bite or octet anywhere in the 32 bit IP number of also produce 0 to 255 and decimal or 80's to 8 ones in binary so be all possible values in our example our 1st example to take 123 in decimal and go down the box and see how many of each book column we need to make up 123 so in this example I need 164 so 64 and 32 is 96 and then we take a 16 so that's 96 is 106 and 6 is 112 and I needed 8 which is 112 and 8 which is 120 needed to what is 122 and a one which is 123 so essentially all you do is go down the box and pick out the numbers that make up your

decimal value decimal value can only be to be between 0 to 35 is to to the 8th produced 256 possibilities were 0 through 255 as I previously mentioned that's all it is is is take a decimal number of pick it out while you need 0 or one in each column and now we have the equivalent of 123 and binary with an example to example 2 a week to take 192 and and in this will rework the binary value and what we find out is 192 is a 128 and there's at least one 11 28 in 192 we subtracted I wanted to for 128 Lisa 64 is a least 164 and 64 and that's it really also 0 there is nothing left of the decimal number and 128 64 = 192 RR binary equivalent in this case is 11000000 this is the primary tool that makes the process so easy and no math is involved so if you're not good with exponents this might be the weighted to to convert your decimal and convert your binary you can also use the calculator available on your PC and use the binary and decimal option key to switch between numbering systems note a small trick really the Rosa binary to the boxwood jazz efficiency during conversion is that the calm to the right is always half the previous comp in this case the bow to the right of 128 is 60 4/2 of half of a 128 is always thirsty for a half is 8430 to half of 32 is 16 and he could see the pattern all the way down a small trick of the calm to the right of the calm is filled with with ones it will be one lesson that column meaning if it were my arrow is on the 64 if all 3216 8421 were all filled with ones that would equal 63 another small trick so for example when every thing to the right is 64 is ones and then the total decimal value 63 which I just mentioned



if you have a subnet mask address the box produces the following results so you're working with subnet mask we can use the same box to compute the binary equivalent to what I see here is if I

have a binary mask of 128 and 128 sits in that 1st position so that becomes 130's and 4 zeros your descendent 1st position if you have a 192 is for the 1st 2 boxes become binary values if you have a 255 then what happens you will produce 8 ones you can also use the magic box a compute out the decimal and binary equivalents resolve that mass how you will now quickly out of the numbers across the top and left and that's your work from left to right on this and this is called adding high order bits in the RFC

Binary

2^7 = 128	2^6 = 64	2^5 = 32	2^4 = 16	2^3 = 8	2^2 = 4	2^1 = 2	2^0 = 1	Decimal
0	0	0	0	0	0	0	1	1
0	0	0	0	0	0	1	0	2
0	0	0	0	0	1	0	0	4
0	0	0	0	1	0	0	0	8
0	0	0	1	0	0	0	0	16
0	0	1	0	0	0	0	0	32
0	1	0	0	0	0	0	0	64
1	0	0	0	0	0	0	0	128

here we have a binary chart showing conversions of each of the columns from 1 to 128 our 1st column 2 the 0 equals one we follow that down onto our 1st row in every else's 0 out then it just the one that would equal 1 if it is using the 2 the 1st thing that is a value of 2 if we follow that down and you follow this pattern over 2 the 2nd which is going to equal 4 if all that down and crawl in a cross-section there is a that's an 8 2 the 4th which is 16 in 2 the 5th which is 32 and 2 to the 6 with a 64 and 2 to 7 June 128 and I could see each one of these decimal values which is mixed up the chart at the top but you can also see them in the binary representation

2^7 = 128	2^6 = 64	2^5 = 32	2^4 = 16	2^3 = 8	2^2 = 4	2^1 = 2	2^0 = 1	Decimal
0	0	0	0	0	0	0	0	0
0	0	0	0	1	0	1	0	10
1	0	1	0	1	1	0	0	172
1	1	0	0	0	0	0	0	192
1	1	1	1	1	1	1	1	255

another binary chart showing the conversion of RFC 1918 prefixes of these are the prefixes that are not used on the Internet they are set aside for private networks they are set aside for network address translation so hopefully you remember the importance of the RFC 1918 prefixes for the 1st one which is the gateway of last resort which is a 0 at the pretty easy conversion 10 is I need an 8 and a2 or 2 to 3rd and 2 the 1st column and it looks like a binary 172 that's what that looks like an binary 10101100 192 we saw that earlier in one of our examples where just the 11 in fall by 6 zeros 255 which is the broadcast for everybody and anyone this is not a direct broadcast is a regular broadcast notice that it's all ones

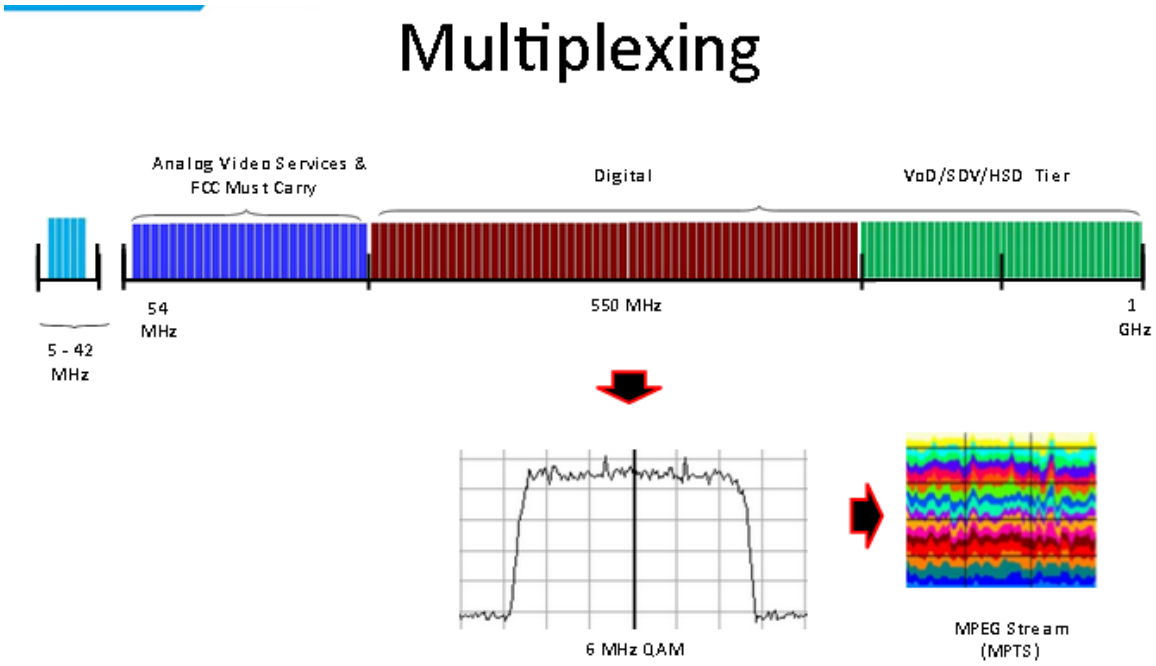
now it's time to pause the training given exercise to convert decimal and binary values on the left I have 5 bullets the 1st 2 bolts are binary values which are to convert them into decimal Genghis Paul is the training the bottom the screen you can visit the pause button get some paper calculate out what would be the decimal values for the 1st 2 binary that I gave you 3 decimal values and what you can compute the binary equivalent to those so take a moment figure all this out and when you're ready you can compare your answer to the notes that he does go into the notes of the training see how you did

Convert the following numbers:
1010 1000
0011 1111
189
74
257

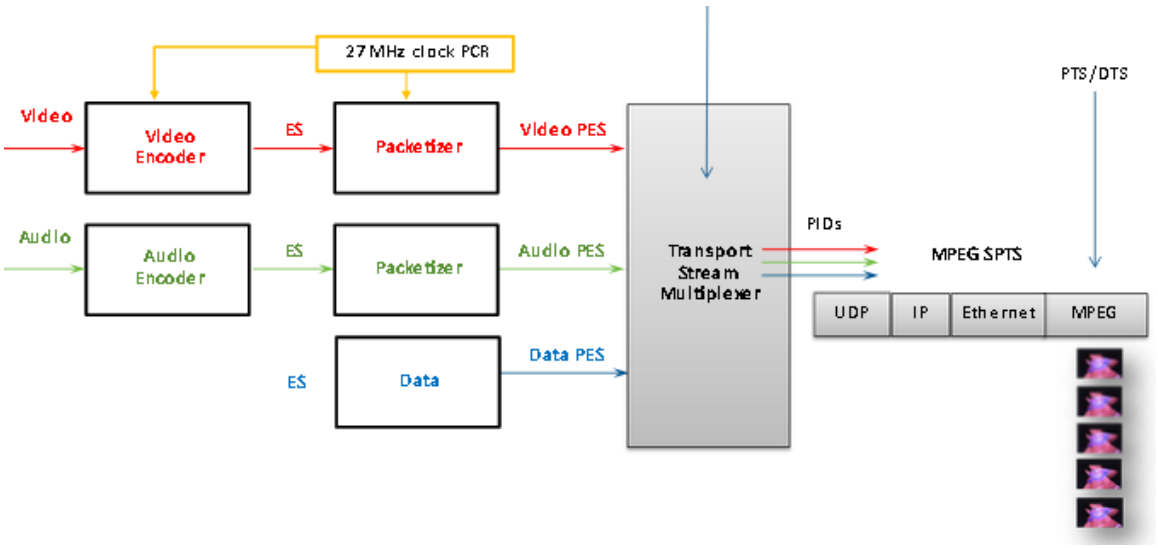
Answers:
1010 1000 = 168
0011 1111 = 63
189 = 1011 1101
74 = 0100 1010
257 = invalid octet or 0001 0000 0001

in this session will explore how digital signals are multiplexed together to form a single channel

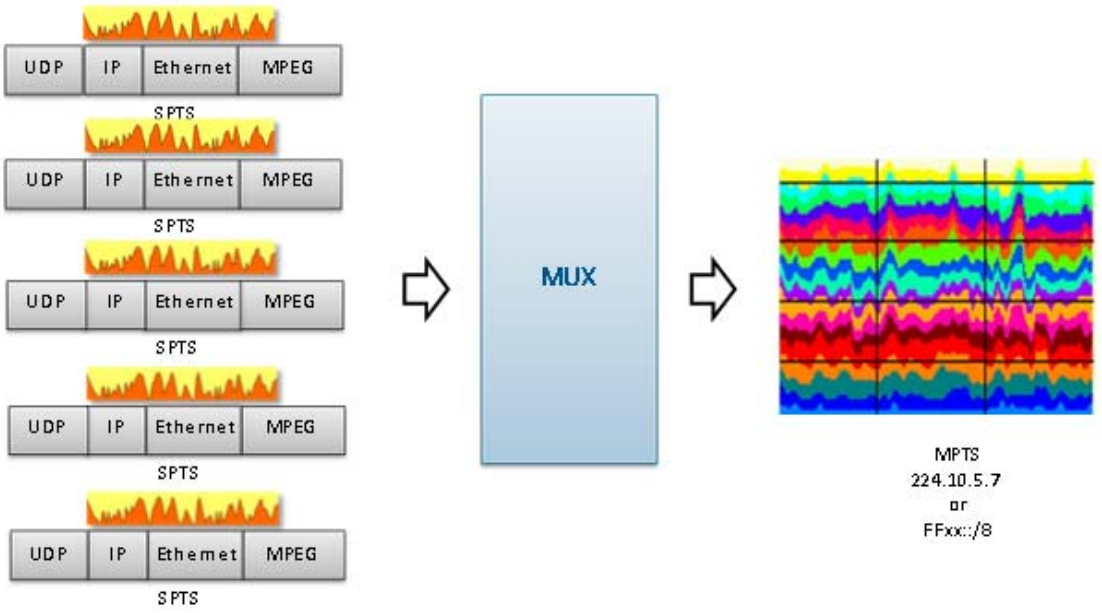
and a 6 MHz bandwidth digital channels are created when data from several video sources is modulated onto a digital carrier signal that occupies 6 MHz of RF bandwidth this is referred to as multiplexing a multiplex of digital video channels is called a multi program transport stream or MPTS if a system offers 48 digital channels in each 6 MHz digital carrier can handle 8 digital video sources then 6 6 MHz channels are required for this multiplex to devices know which data belongs to which service when a customer chooses specific program to watch the processor within a digital set-top box directs the unit to the appropriate frequency program identification or PIDs are information that's encoded within the data for each channels bitstream the set-top box device looks for data this contains the specific program identifiers



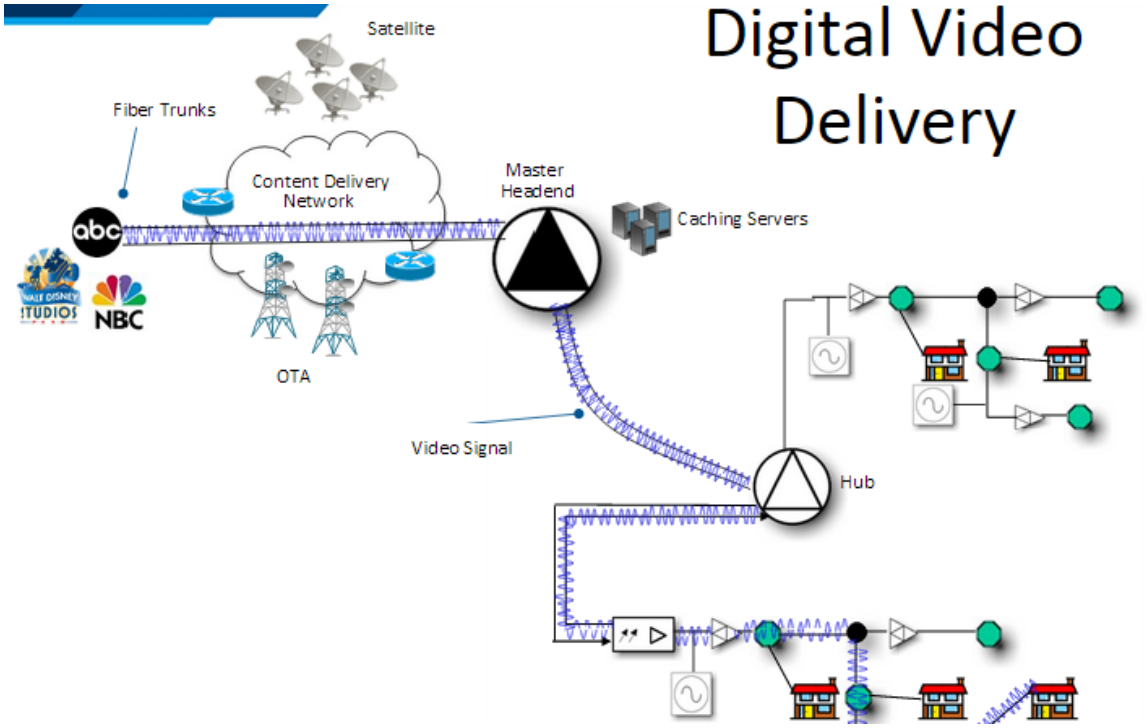
multiplexing combines several signals into a single signal allowing easy transmission over the cable network the forward path uses a frequency division multiple access FDM to break up the bandwidth the 6 MHz chunk shown here everywhere from 54 MHz all the way to 1 GHz at everything is a 6 MHz chunk channel time division multiple access TDMAis using return paths separate upstream signals using timeslots ATDMA SCDMA and OFDM or other options for return paths multiplexing more information on multiplexing techniques is coming up in a future module just understand for now that the forward and reverse path uses technology to multiplex signals over the outside plant in the forward path each 6 MHz channel is further multiplexed using moving pictures experts group or MPEG and in MPEG we use different transport streams so you could have a single digital program which would be a single program transport stream SPTS per 6 MHz or more common a multiple program transport stream MPTS per 6 MHz channel



an MPEG single program transport stream er SPTS is created from elementary streams or an ES an ES is the most basic component of MPEG is created by encoding video and audio as shown here a packetizer is used to convert each raw video and audio ES into a pocketized ES or what we call a PES or PES that may be transmitted over a cable IP or docsis network each PES packet I will be a variable number of bytes in length up the 65,536 bytes per block and it also include a 6 byte protocol header each PES packet header includes an 8 bitstream ID identifying the source of the payload which we just talked about calling the PID in addition to the creation of the packetized video and audio a data packet that in some examples a control data may be added to the packetized stage. so we see down there in blue we have a small data box adding some some data to the to the single program transport stream each packet size elementary stream or PES is then synchronized for a common 27 MHz program clock reference it as is for reliable transmission of the cable network the PCR is added to the presentation and decoding timestamp feels as well in the MPEG frame the PTS will be use of the time which a decoded audio or video access unit is to be presented by the decoder a DTS is the time at which an access unit is decoded by another decoder each PES is multiplexed into a single program transport stream or SPTS and that as SPTS is packaged into MPEG transport stream with a PMT table or program map table because as indicators on how to pull out the individual A/V data pieces a transport stream with more than one program is called a multi-program transport stream the PMT contains a set of program IDs or pid values of each PES which is the video audio data which comprises a television program or SPTS



multiple single program transport streams will be combined and that will be combined into a multiple program transport stream MPTS shown here each as SPTS must be encapsulated in ethernet with a unique IP address in the case of of our example a IP address of multicast type will be used and that's typically a 224 address or in the IPv6 and FF address and also will use the datagram user datagram protocol UDP to assign a port specifically for this video transport so there are several layers involved in the multiprogram transport stream in the single program transport stream ethernet IP and UDP



now let's focus our attention on digital signals over the HFC or outside plant here's an example of the modern cable operator digital video delivery network fiber trunks are used to connect major networks to the master headend or content delivery network CDN shown up in the left upper left-hand corner over the year and 10 is mounted on towers are use receive digital video signals at the master headend for over the air signals caching servers use of the headend hub sites to cash digital video content for the subscriber video signals are transmitted over the HFC use and MPEG transport and docsis during the distributing of digital video signals they are susceptible to the same degradation as analogs we have are digital signals coming in the network at the hub and as those signals move out of the hub there to be back in analog mode because her going over fiber optic digital video data travels through various devices and the cable network that increase the signal level so you can see we get to the amplifier how that signal becomes changed a bit

a digital signals transmit using the same fiber collects cables as analog signals the digital signals are separated by frequency known as the CEA542C the hybrid fiber coag surrounds I plan hardware is an analog transmission path treating analog and digital signals the same

digital and analog signals need a carrier to be transmitted over the HFC are outside plant the carrier provides constant energy at specific frequency for transmission over the HFC or outside plant modulation is the process of applying cable information onto the carrier's specific bandwidth saying 6 MHz in our case, digital signals that use the HFC or outside plant are high-speed data voice over IP video-on-demand switches will video standard high-definition 3-D and ultra definition video channels

digital signals in HFC or outside plant have specific metrics to determine the health of the signals

as they are transmitted over the HFC a few common measurements are modulation error ratio or MER error vector magnitude are EVM and bit error ratio BER

digital signals over the HFC are outside plant produce and are impaired by distortions noise and interference when digital signals leak power outside the allocated bandwidth it can produce a noise known as distortion interfering with other signals in the HFC or outside plant there are distortions that are in channel and out of channel noise can be a network distortion random signal energy added to the digital signal by active devices in HFC outside plant

ingress or interference is noise or distortion them and make their way into the also plan from sources outside the system analog signals may still convey video information when they are extremely distorted in digital this is not the case digital maintains a higher level of quality than analog and when exposed to the same distortions of digital signals fail completely as shown here

there are several technologies of the assist will signals from complete failure digital signal processing the process of encoding analog into digital the signal compression and error correction to help conceal the digital signal adaptive equalization that may be used to step down the quality of a digital signal for example in a digital video digital signal there are several bandwidth steps available to back down the quality of the signal when being with his tight

a high level look at digital signal modulation a deeper dive of modulation is in a future module

there are 2 categories of modulation available analog and digital cable operator moving away from analog modulation techniques shown on the left a carrier signal or continuous wave contains no information of its own we just learned that the provided energy at specific stable frequency to transport digital signals this enables reliable signal transmission the process of adding intelligence or information from a baseband signal to the carrier by changing the carrier signal frequency amplitude phase or some combination of these characteristics is called modulation the frequency band surrounding each carry is also known as the transmission bandwidth the carrier of modulation must remain within the allocated bandwidth of their are good to keep from being corrupted or causing interference to other signals

using modulation to transmit information falls under 4 main steps the 1st step in operator will do is a generated pure RF carrier at the headend transmitter at a particular frequency the 2nd the pure RF carrier is modulated within with the information with others video data voice etc. to be transmitted to the customer and there's examples on how signals are changed and we can do it with amplitude frequency phase or combination of amplitude and phase called Qam the modulated signal in step 3 is combined with other signals at the headend and fed into the laser transmitter and and and last at the receiver in the field the modulated signal is detected and demodulated recover the information

a more advanced phase shift keying is multi bit modulation quadrature phase shift keying QPSK employees periodically shifting the phase of to in phase carriers I and Q at 600 Bd or 90° shift plus encoding technique QPSK calls for 4 define phase changes or angles to represent data symbols showed here 0 90 180 into 270 is a 90° separation between the phase angles and one phase shift per period of the wave means that each. Can represents 2 binary data bits per symbol since this is possible that makes QPSK 2 bits per symbol in QPSK basically it's is to raise

the power of to give you 4 different symbols shown here in the note shall find a different symbol values for QPSK in legacy more information and more details coming up on modulation in the future in the near future module but QPS case typically use any upstream by cable operators for slower data services such as out of band signaling to set-top boxes is also a very resilient for satellite signals in the headend or hub site because it produces an it has a lower signal noise but produces a good clean signal QPSK operates well in a noisy upstream offers low data rate as compared to Qam

here is the QAM signal you're probably all familiar with or called the haystack and this is how it would appear on a spectrum analyzer the horizontal axis shows frequency and the vertical axis shows amplitude many different RF modulation schemes can coexist on a coax or HFC network a form of a.m. is used for analog video channels FM may be used for broadcast if broadcasting it is carried over the same network some older set-top converters may use the BPSK for data communication QPSK is sometimes used for telephony services Qam is used for data services and digital video services the choice of modulation scheme depends on the type of signal to be transmitted and the cost and the RF bandwidth being with required size we start using QAM palms will provide more bandwidth than QPSK and here is the 1st QAM is called 16 QAM and were representing it in 2 different ways one by the constellation diagram on the left and one by the spectrum view on the right 16 QAM is 4 bits per hertz or 2 bits on the i Chat on 2 bits on the Q were 4 bits per symbol shown here in a gives you a total of 16 total states or 2 raise the power f4 16 QAM is typically use any upstream return Path for data services and newer set-top boxes boxes specifies that the operator can select an upstream modulation of using QPSK which is better in noise because it offers much bandwidth or you can use a Qam on the screen is the I and Q constellation plot showing 4 levels of I and 4th levels of Q

to increase at a rate operator can choose a higher order modulation technique such as 64, 64 QAM's were symbols are equal to 6 hertz or 3 because of I and 3 bits of Q 6 bits per symbol yield 0 to 63 possibilities or 64 total states or 2 raise of the power of 6 64 Qam is typically used in the upstream return Path for data services and new set-top boxes some really 64 QAM was selected for the downstream were Forward path back in the day but now 256 QAM is more common boxes specifies that operator can select upstream modulations of QPSK or QAM to continue to further increase the data rate over 64 QAM in operator can choose 256 QAM requires the least amount of RF bandwidth for a given amount of data 256 QAM's were symbols are equal to 8 bits per hertz 4 bits of the I and 4 bits of the Q 8 bits per symbol yields 0 to 255 possibilities are 256 total states or 2 raise the power of 8 256 QAM's typically used in the downstream were forward path linear video and data services an operator can select different QAM downstream channels

for the forward path 1024 QAM has been shown feasible in a clean spectrum so this provide even greater bandwidth for video signals also to clean spectrum 4096 QAM has also been shown feasible in Europe using digital video broadcast over cable 2 which is the DVB-C2 to in the service as a 50% increase in the capacity over 256 QAM a signal to noise of 46 db or greater is required to achieve 4096

2.

the 1st section will explain the fundamentals of analog signals national television systems

committee or NTSC is the analog television system in use in United States Canada Japan Mexico the Philippines South Korea Taiwan and other countries it is named United States standardization body that adopted it the 1st black-and-white monochrome NTSC standard for broadcast was developed prior to the second world war and had no provision for color transmission the standard called for 525 lines of picture information of which 480 reusable in each frame and 29.97 frames per second the frame rate was later slightly adjusted for the color standard and that's this 29.97 civilian development of commercial television was halted with the entry of United States in the war in 1953 a second standard was was issued which allow for color broadcasting to be compatible with the existing stock of black-and-white receivers while maintaining the broadcast channel bandwidth already in use this was an important commercial advantage over incompatible color systems that had no at that have been proposed at this time NTSC was the 1st widely adopted broadcast color system after over a half a century of use over the air NTSC transmissions are now being replaced with advanced television systems committee ATSCover the air digital transmissions in the United States analog transmissions were halted in 2009

NTSC means 4 units by 3 units dimensions originally chosen to match 35mm motion picture film or 1.33 to 1 aspect ratio wider aspect material must be compressed for viewing on 4 x 3 sets using a pan and scan a letterbox video or anamorphic squeezing the full resolution of NTSC is 720 x 525

Analog Regulation

- Countries have a frequency allocation map which assigns specific RF frequencies to VHF and UHF channels
  - VHF - 30 MHz to 300 MHz
  - UHF - 300 MHz to 3 GHz
- Cable operators required to follow a frequency map
- PAL and NTSC

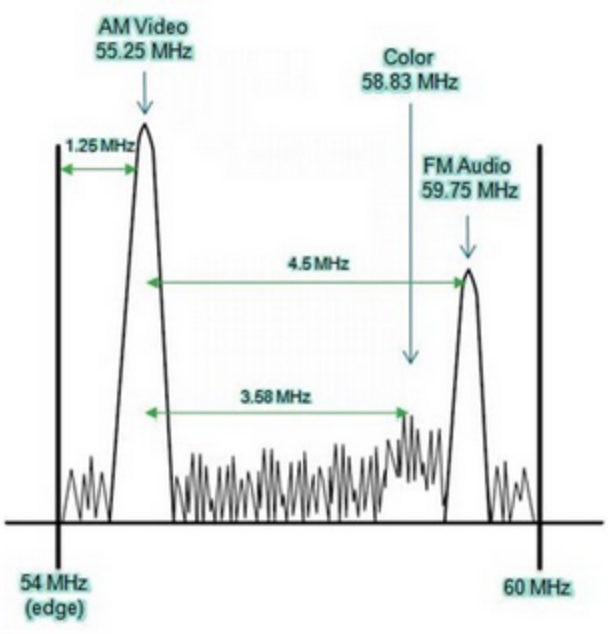
Channel Number	Off-Air Frequency	Cable Frequency
2	55.25	55.25
3	61.25	61.25
22	519.25	169.25
23	525.25	217.25

over the years the FCC is created a frequency allocation map which assign specific RF frequencies to the international telecommunication Union's very high frequency or 30 MHz to 300 MHz and ultra high frequency UHF or 300 MHz to 3 GHz channels the map is a

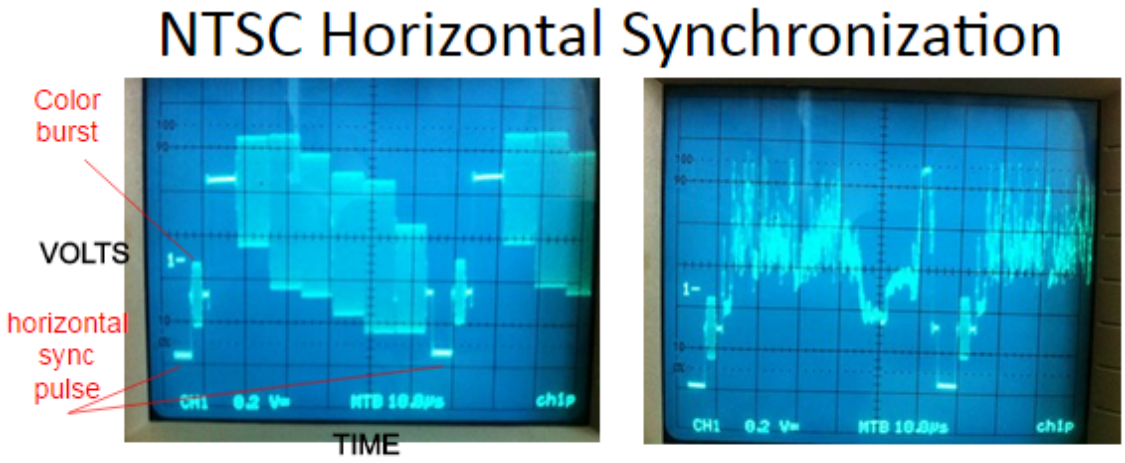
nonnegotiable and all broadcasters and manufacturers must follow to ensure compatibility and to avoid interference from channel to channel on all equipment at different points throughout the years a federal authorities such as the FCC has also allocated frequency frequency map to cable operators and multisystem operators because this map was implemented through different periods and because lower frequencies carry better on coax the cable map does not match the off air map for all channels in the channel cable channel frequencies are not always in order a channel is defined as 6 MHz of space or 8 MHz in Europe it takes a full 6 MHz to broadcast one analog video am modulation with synchronized audio fm modulation the national television standards committee or NTSC in the United States or phase alternating lines in Argentina set the color or set the channel size

the NTSC system use a horizontal scan rate of 15.75 kHz horizontal scan rate is the rate at which each line of video is scanned from an image and is reproduced on a television receiver in the monochrome system the field rate is 60 Hz resulting in a frame rate around 30 Hz is initially this initial selection of field rate was chosen in order to lock the frame rate of early televisions to the 60 Hz power means the NTSC system was interlaced scanning each frame of video contains 525 lines there are equally divided between field one and field 2 each field having 262.5 lines of the 525 lines only 480 lines are used for active video information the other lines are used for syncronization and other purposes consequently NTSC is often referred to as 480 I were the I is standing for interlaced the monochrome system was subsequently modified to create the compatible color system sometimes referred to as NTSC2 to frequencies were adjusted slightly to minimize interaction between luminance signal and the chrominance signal in this system the horizontal scan rate is 15.734 kHz and the field rate is 59.94 Hz and the color carrier frequency is approximately 3.58 MHz these choices of scanning field rates are sufficiently chosen to the monochrome frequencies to be compatible with monochrome receivers occasionally a video signal may come from a source such as a videotape recorder that does not produce frequencies perfectly comply with the NTSC standard a device known as a time-based corrector may be used to digitally correct these errors make the timing of the signal practically compatible with the NTSC signal there are time-based correctors for all video standards as well

NTSC / SMPTE 170M



national television systems committee 6 MHz RF standardize channel shown here as an example channel 2 exist within the bandwidth of 54 MHz to 60 MHz the signal components are video at 55.25 MHz which is 1.25 MHz from the edge color at 58.83 MHz which is 3.58 MHz from the video and audio at 59.75 MHz which is 4.5 MHz from the video this SMPTE society motion pictures and television engineers they have a standard NTSC composite analog television signal known as 170m the overall level of composite NTSC video signal is a 1 V peak to peak



- Horizontal scan rate 15.734 kHz means each line of video lasts for 63.557 microseconds
- Horizontal sync pulse lasts for 4.7 microseconds
- The color burst consists of 9 cycles of the 3.58 MHz color carrier

NTSC color television consists of horizontal scans of information that occur at 15.734 kHz rate which we talked about this means that each line the video last for about 63.557 µs microseconds that period consists of approximately 52 µs of active video information the rest is horizontal syncronization information this horizontal syncronization information is used to tell a television receiver to begin the display of each line of video the horizontal syncronization information occurs during a part of the video waveform known as the horizontal blanking interval as the name implies the heart of the waveform is blanked from display on the screen

shown here are 2 samples of an NTSC video each trace displays approximately 1 1/2 lines of video that the horizontal resolution is 10 µs per division in the vertical resolution is .2 V per division horizontal sync pulse can be seen on the lower left and again approximate two thirds of the way across the screen both the horizontal sync pulses the front porch are followed by the back porch containing the color burst the left race is video for a series of color bars the chrominance signal riding on each level of the luminance indicating the color information for each bar the waveform for the right is a random sample of lines of active video the NTSC horizontal sync 50% point is used to synchronize digital sampling

we just talked about horizontal synchronizing let's talk about vertical synchronizing vertical synchronizing information is contained at the end or beginning of each field of video occurs during the vertical blanking interval or VBI the vertical blanking interval begins when a series of negative going pulses that are half the width of the normal horizontal sync pulses that occur at a

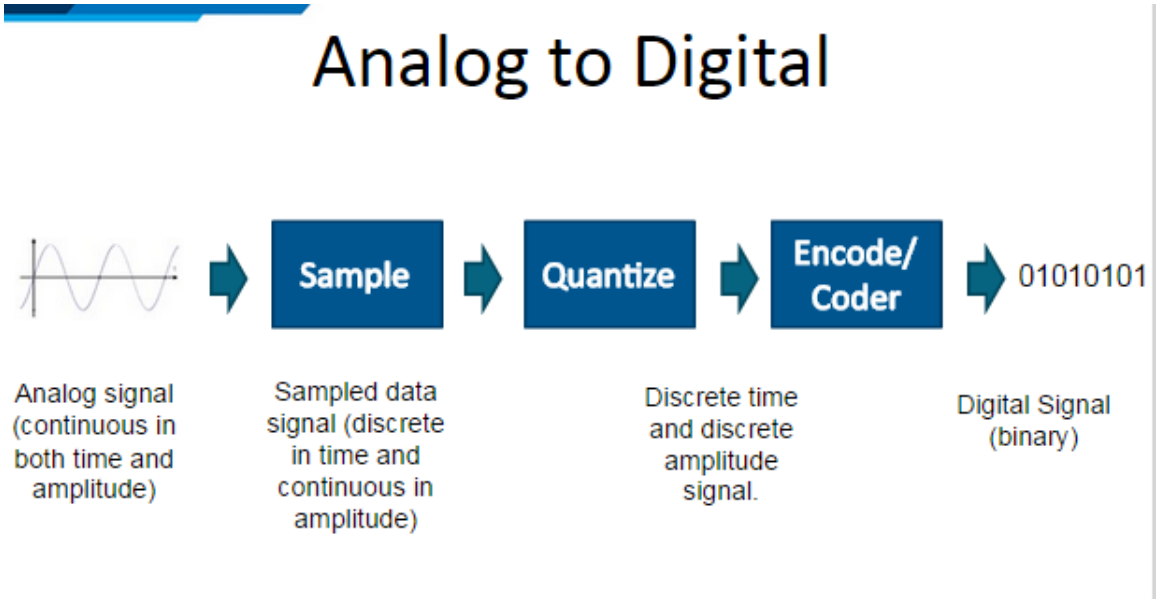
rate of 2 per line of video these narrow pulses are referred to as equalizing pulses the 1st group of equalizing pulses is followed by a series of 3 lines were the signal is at sink level for all except 4.7 µs at the beginning and the middle of each line these wide pulses are referred to as vertical sync pulses following the vertical sync pulses are a second series of equalizing pulses excact number of and position of equalizing pulses identify which field is about to be transmitted the 1st vertical blanking interval begins with a half line and ends with the half line shown here is a vertical blanking interval followed by 7 lines of video note that the horizontal time base is 100 µs per division note the series of equalizing pulses followed by the vertical sync pulses intern followed by the second series of equalizing pulses

the time-based corrector video processor allows for color correction and frame synchronization in NTSC as seen in a previous illustration lines 10 through 20 are normal lines with horizontal sync pulses and color bursts but do not contain active video information these lines occur outside the visible picture area in a properly adjusted television receiver lines 17 and 18 are typically used for vertical interval test signal or VITS line 19 may be used for vertical interval reference signal or VIRS line 21 may contain active video or may contain close captioning data CEA-608 and EIA 608

A/79 ATSC conversion to NTSC provides guidance to broadcasters and other creators of ATSC high definition and standard definition content and to operators of multichannel video programming distributor systems a79 8 ATSC conversion to NTSC recommends the equipment has given abilities needed to provide the highest quality programming to viewers who only receive NTSC services

the phase alternating line or PAL system was developed after the NTSC system in this system the phase of the chrominance signal including the color bursts changes by 90° on alternating lines this change is made to address issues with color hue and saturation resulting from gain and phase errors in the transmission system the PAL television sets typically do not require a hue adjustment the PAL aspect ratio like NTSC is 4 x 3 the PAL system uses 25 interlaced frames as opposed to 29.97 in NTSC 25 interlaced frames is also 50 fields per second or 50 Hz each frame is made up of 625 lines of which 576 are used for active picture information system is often referred to as 576 I were the I stands for interlaced several versions of PAL systems exist each was a different color or color carrier frequency and oral carrier frequency the PAL system is used in much of Europe Asia Australia and South America

the French system sequential color aim and more or SECAM is intended to eliminate some of the picture artifacts that result from top simultaneous transmission of the luminance signal and the I and Q color signals one of these artifacts is typically referred to as dot crawl the SECAM system alternately transmits the I and Q color signal on alternate lines color information is frequency modulated onto a color carrier memory and the receiver stores the information from the previous line to a combined and present line to re-create the color information as a consequence the SECAM has only one half of the vertical call resolution of the NTSC system SECAM white PAL is a 576 I system there are 625 lines per frame and 25 interlaced frames per second or 50 fields per second providing 50 Hz



analog-to-digital or A to D or analog to digital converter ADC the equipment used is known as a coder decoder or codec coder converts analog signals to digital binary bits the decoder converts digital binary to its analog signals or sine waves the process the A/D process is made up of sampling quantization and encoding then that determines the quality of the digitized signal transmission of bits with equal gaps is called iso chronic transmission we cannot have uneven gaps between frames so analog to digital conversion we see on the screen here we've an analog signal on the left-hand side to continuous in both the time and amplitude we sample the data discrete in time and continuous amplitude and we quantize that we reduce the samples and that becomes a discrete time in a discrete amplitude signal and then we take the quantize values and encode them and we now we we get is binary data or a digital signal

a Bell Labs scientist Harry Nyquist produced a mathematical proof of the 1920s the proof that the full information content of any continuous analog baseband signal can be recovered at the siganl measured encoded at a rate that is twice the frequency 2 times F Max of the highest sinusoidal component contained in the analog signal nyquist sampling is the 1st step in the conversion process the Nyquist sampling rate is ideal for low pass analog baseband signals the sampling may occur before modulation and multiplex and or after Demultiplexing and modulation the composite video signal created by devices such as game consoles and DVD players commonly use baseband signals

Nyquist theorem examples let us sample a simple sine wave at 3 sampling rate you have the frequency sample which is FS equals 2 times the frequency which is the Nyquist rate you also have frequency oversampling which is FS equals 4 times a frequency or 2 times a Nyquist rate or you have frequency under sampling FS equals frequency half to Nyquist rate you can clearly see that we have redundancy built in the 4 times frequency we don't have enough in the half frequency so it can be seen that sampling at the Nyquist rate can be can create a good approximation of the original sine wave oversampling can create the same approximation is

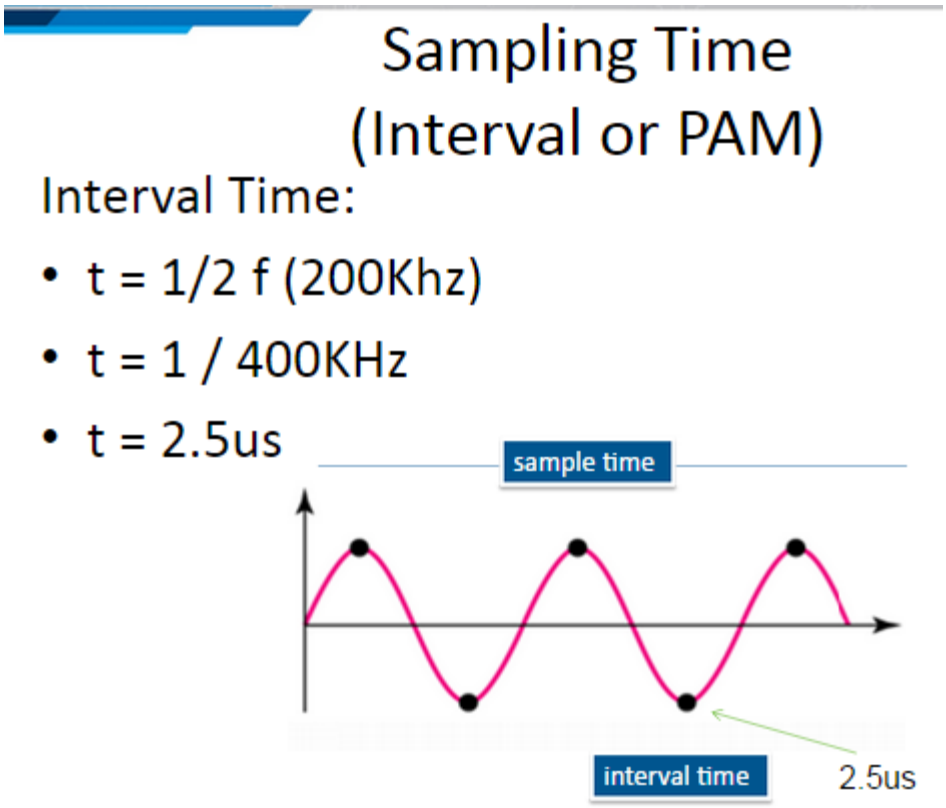
redundant and provides extra data that the operators don't need sample below the Nyquist rate does not produce a signal that looks like the original sine wave example of a digitization process for voice signals and speech for Cable phone by assuming a maximum frequency of 4000 Hz or 4 kHz the sampling rate for digital phone FS equal 2 times a frequency therefore we should take 8000 samples per second to encode the digital voice signals

considered the revolutions of a hand on a clock the second hand of a clock has appeared as 60 seconds according to the Nyquist theorem we need to sample the hand every 30 seconds this is called time sample board T sub S which equals time times half the receiver of the samples cannot tell the clock is moving forward or backward

in this example we oversample at double the Nyquist rate or every 15 seconds T times a quarter the sampling are 12 369 and 12 the challenge with this way that the sample shows a clock moving forward

here we under sample at half the Nyquist rate or every 45 seconds T times 3 three quarters the sample points are 12 9 6 3 in 12 although the clock is moving forward the receiver thinks of the clock is moving backward related example is the seemingly backward rotation of wheels of a forward moving car in a movie this can be explained by undersampling a movie is filmed at 24 frames per second if a wheel rotation is more than 12 times per second the undersampling of 12 frames increase the impression of a backward rotation in the video

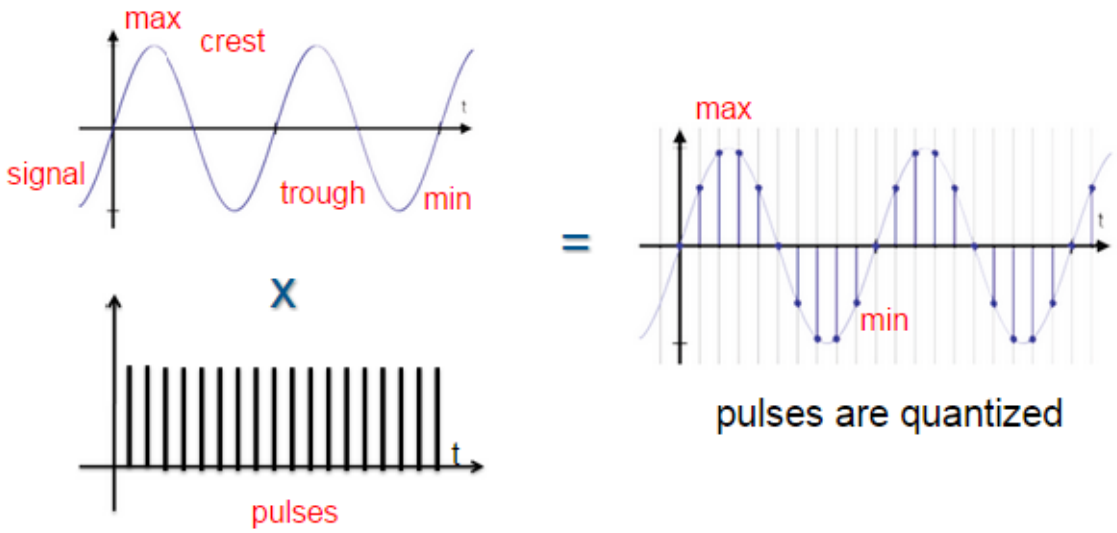
a complex baseband signal has a bandwidth of 200 kHz what is the minimum NyQuil sampling rate for the signal and you could pause your training here the answer is in the notes



notice that the time to take the sample is significantly less than the interval time if the sample time equals the interval time this would be 100% sample at regular intervals therefore a snapshot of the signal being taken is done the sample signals are called pulse amplitude modulation or Pam signals the Pam signal has amplitude of the analog signal at the moment of measurement modulation is a digital or analog signal inside another to be transmitted over transport medium a Pam messages encoded in the amplitude of a series of pulses a microsecond is abbreviated US or 10 times -6 and here is our example again for interval time

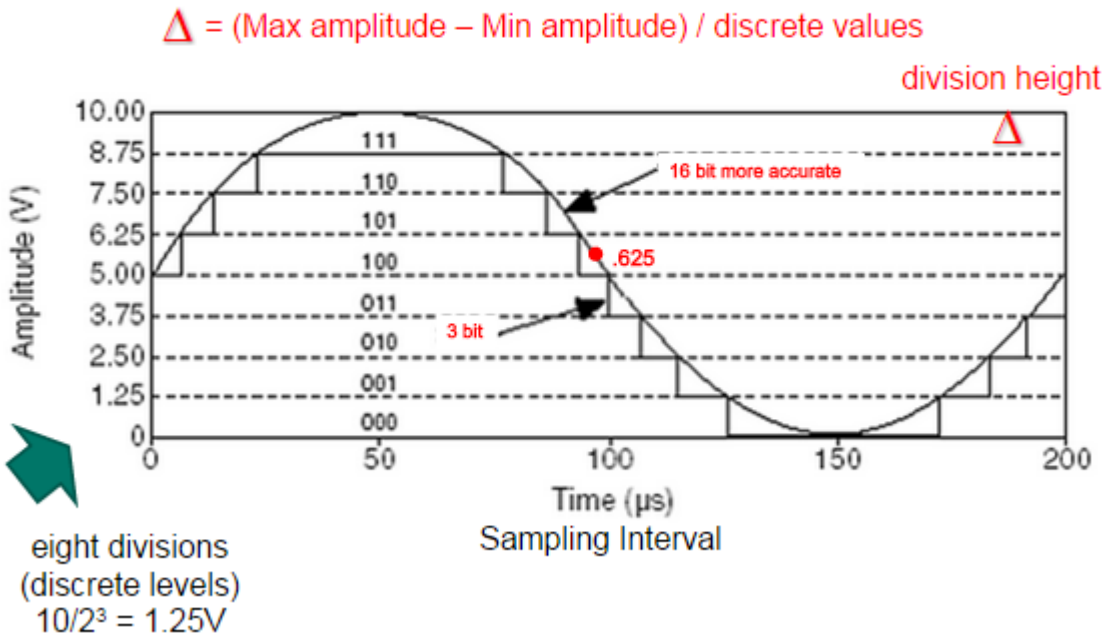
Quantization

Pulses are quantized (mapped) to some finite number of quantization levels (known values).

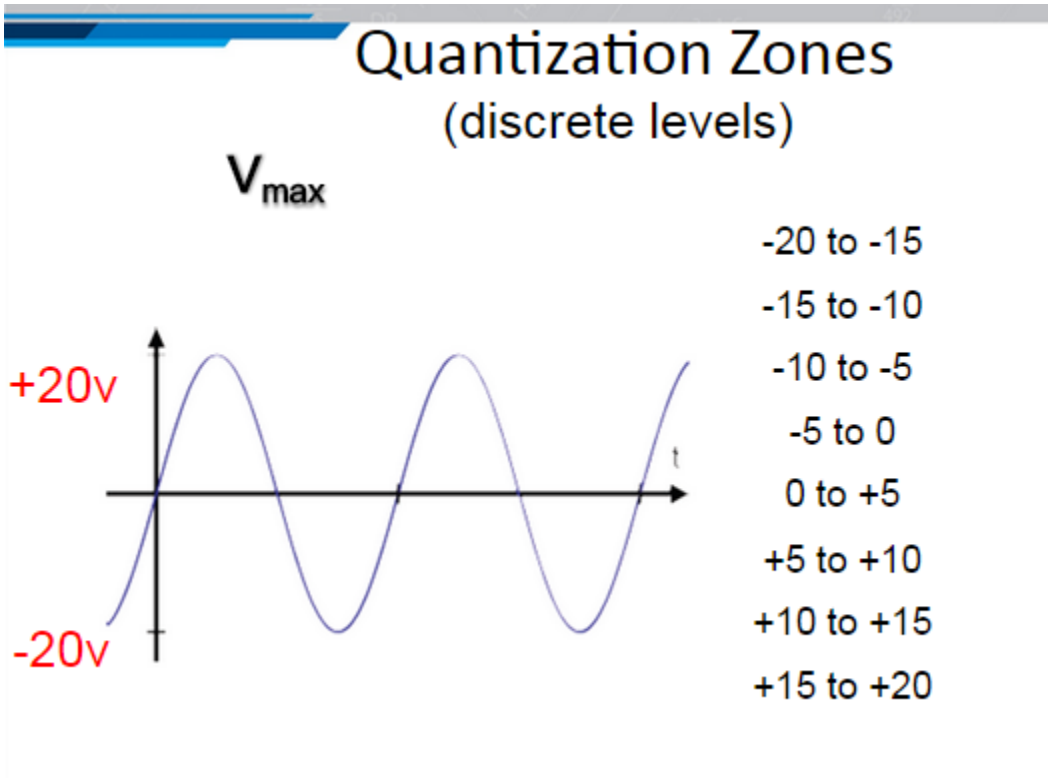


once we have our samples quantizing the Pam signal is the next step establishing a discrete unique value for each sample based on an estimated set of values quantization is defined as the process of converting an analog signal to a digital representation quantization is performed by an analog-to-digital converter sampling time based on the time results in a series of pulses of varying amplitude values range between 2 limits a min and a Max shown here on the left side of the screen in order to manipulate the signal with a microprocessor or microcontroller these pulses have to be converted to a binary or digital code before this conversion the pulses are quantized map to some finite number of quantization levels notice values this sample has a value amplitude which must be recorded and kept track of so that the original signal can be recovered

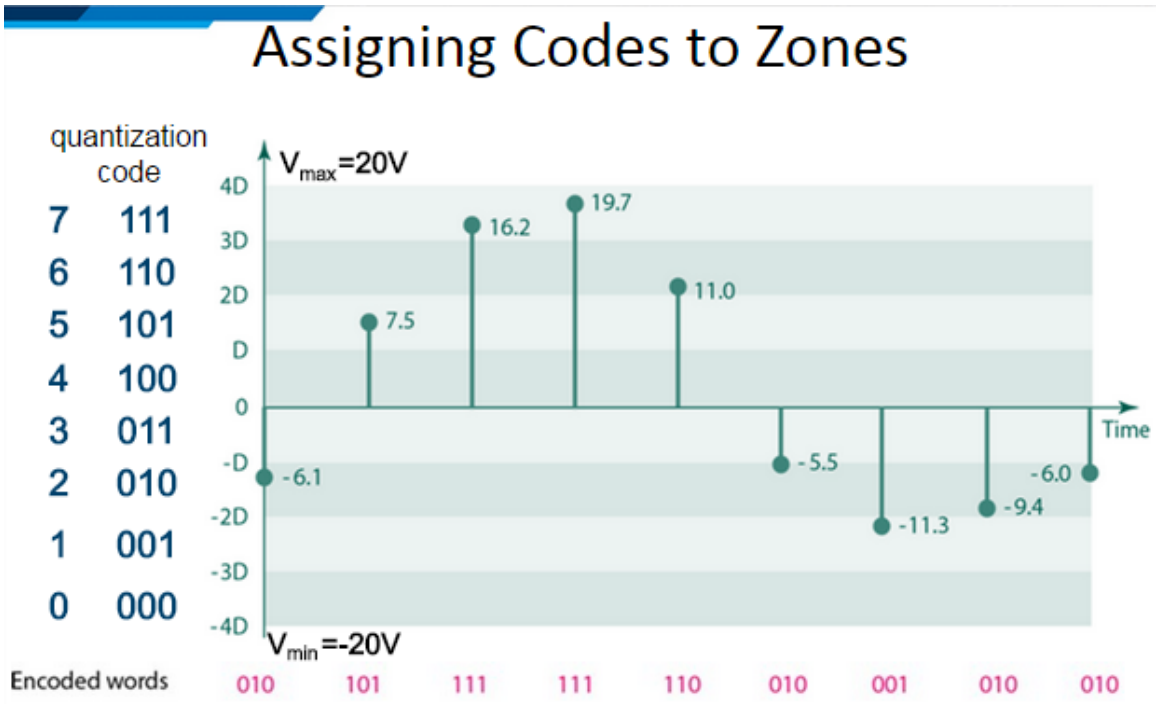
Quantization



a step quantizer or is used in converting analog-to-digital here is a digital image of a 5 kHz sine wave obtained by a 3 bit analog-to-digital converter a 3 bit analog digital converter devide a distance between the min and Max by using binary 2 to the 3rd to the 3rd produces 8 different divisions if we were doing a 4 bit analog to digital converter it be 2 to the 4th or 16 divisions these divisions are discrete levels or L zones each division has a height the height equals a Max minus min divided by the level or in our case 10-0 divided by the 8 or 1.25 jumps and are indicated on the amplitude level and the left-hand side increasing the resolution is 16 bits to increase the analog-to-digital converter number of divisions from 8 to 16 bits which you be 65,536 levels allows a 16 bit analog visible converter to obtain an extremely accurate representation of analog signal but you have a larger signal at the end the idea you were trying to reduce of the size as we go through this process to but so the operators of the play between quality versus size the midpoint of each zone is assigned a value from 0 to L1 resulting in L values E sample following in a zone is approximated to a value of a midpoint in this case 1.25÷2 or .625 and noticed that those midpoints equate over to a binary value



assume we have a voltage signal with amplitudes of V max and V min so the min is -20 V and the Max is positive 20 V we want to use a level that equals 8 or 2 to the 3rd quantization levels the zone width is a Delta which equals 20 minus -20 divide by 8 or 40 by 8 or 5 as the zone width as indicated on right side the zones are described here the midpoints of each zone is when we divide those in half so the midpoint zone for 20 and 15 is 17.5 so is half of 5 which is 2.5 subtracted off the 20 are added to the 15

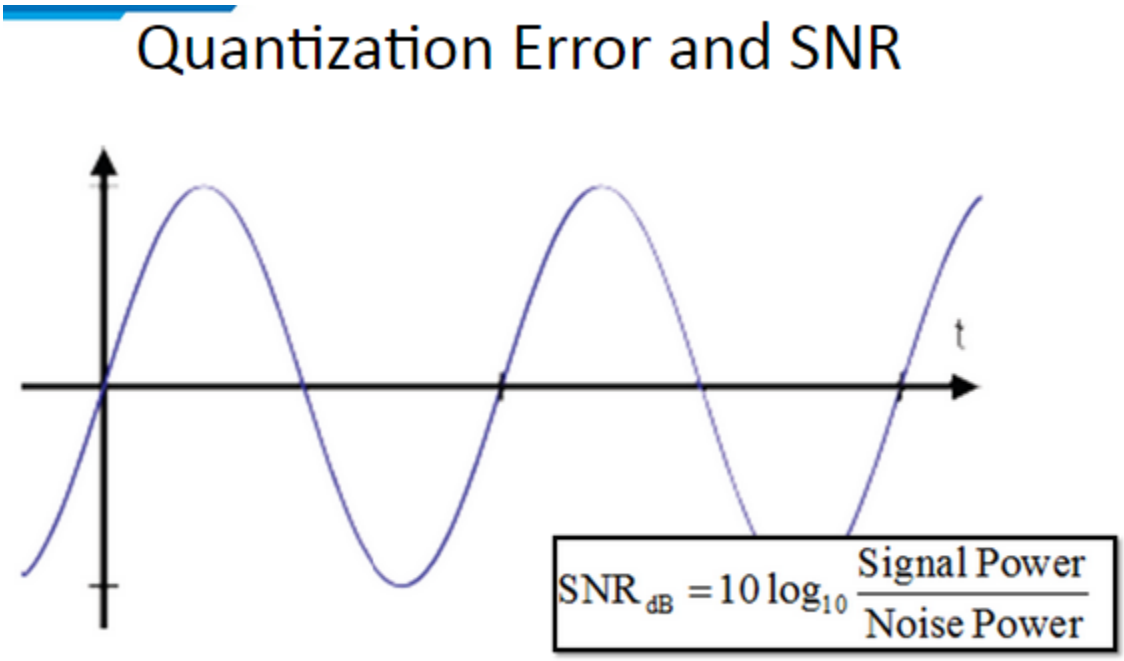


now they were comfortable with the zones in the midpoints the Pam signal is then given a fixed value each zone is assigned a binary quantization code in decimal from 0 to 7 and encoded word from 000 2 111 for a 3 bit analog-to-digital converter the number of bits required to encode the zones or the number of bits per sample as it is, referred to is obtain as follows to give our example are number of bits equals 3 we have 2 to the 3rd which is 8 zones so therefore codes go anywhere from 000 which is a decimal one all at the 111 which is a decimal 7 assigning Codes the zones 000 will refer to zone -22 -15 and 001 to zone -15 to -10 and will keep moving up for each one of the steps

we now have binary from the quantization mapping process the Pam signal will round up or down approximation of the actual amplitude value the fix values depending on the set of raws that are programmed in the headend equipment with the rounding takes effect there are certain amount of incident again errors introduce into the process this is known as quantizing error or distortions the quantization error depends on the number of bits in the converter along with its errors noise and nonlinearities not in a streamline the difference between the actual and coded value the midpoint is referred to as a quantization error or quantization distortion the more zones the smaller the Delta which results in smaller errors but the more zones the more bits required to encode the samples and the higher the bit rate so for example the quantization levels are 0 1 2 3 etc a pulse width magnitude between 1 or 2 will be quantized to either a 1 or 2 this is a fact in this fact introduce noise into the sample signal such noise is known as quantization noise how is problem overcome by setting up values and the algorithm the raw such as the that the error is not detectable to the receiving equipment or humans in other words the amount of quantization error is too small to make a difference

you may pause the training here and got some questions for you think yourself what is a Pam

signal what is quantizing error and why does this happen and when you see the answer does take a look at the notes



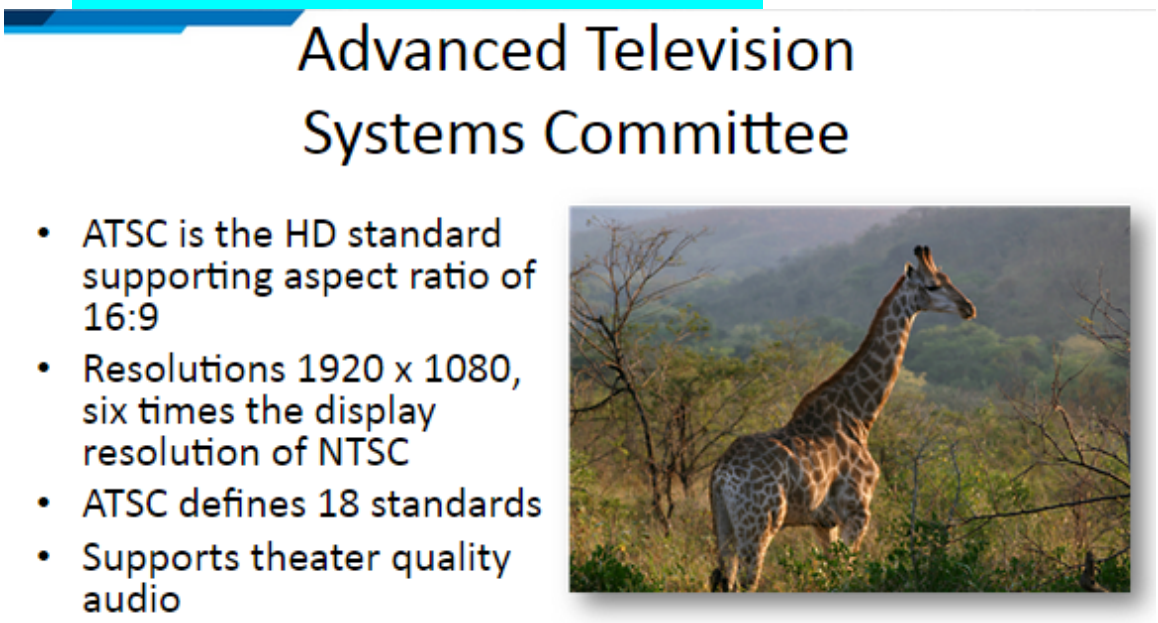
signal to noise ratio of a system is a ratio between the signal power and the noise expressed in decibel or DB signal noise is the average signal power divided by the average quantization noise or noise power indicated here signals with lower ampli values will suffer more from quantization error as the error range the midpoint which equals the Delta divided by 2 is fixed for all signals nonlinear quantization compressed signal is used to alleviate the problem of quantization affects the goal is to keep signal noise fixed for all sample values increase the quantization bit value of the decoder the 2 approaches the quantization levels folly logarithmic curve smaller deltas at the lower able to do larger deltas at the higher Apple tubes you have companding algorithm the sample values are compressed at the center into logarithmic zones and then expanded that the receiver zones are fixed in height

during quantization of the time to mean we could almost completely preserve the waveform information by increasing the sampling interval that we talked about but that creates a higher bit rate in the amplitude domain we can preserve most of the waveform information by dithering dithering involves the deliberate addition of noise to our input signal it helps by smearing out the little differences in amplitude resolution the key is to add random noise in a way that makes a signal bounce back and forth between successive levels of course this is this in itself just makes it signal noisier but the signal smooth is out by averaging the noise digitally once the signal is acquired mathematically averaging the digital signals without dithering does not remove the quantization steps it simply rounds them out a little as shown above

the bit rate of a pulse code modulated pcm signal can be cackling from the number of bits per sample and the sampling rate for the bit rate equals a number of events times the frequency samples suffering for example if we have 3 bits per sample in the frequency was 8000 Hz then I would be 24,000 bits per second the bandwidth required to transmit the signal depends on the type of line encoding used digitized signal will always need more bandwidth than the original analog signal is the price we pay for robustness and other features of digital transmission

let's try one unit again posit the training here and the answer is in the notes that were given a telephone audio levels where there from 300 to 3400 Hz were asking what is the sample frequency needed and if the samples are 8 bits per sample what is the bandwidth

at this point all the bits have been encoded into digital data and sent down to a decoder to recover analog signals from a digitized signal we follow the following steps we see the digital data in this diagram moving into a hold circuit we use a hold circuit that holds the amplitude value or pulse to the next pulse arrives the whole circuit then passes the information through a low pass filter with a cutoff frequency that is equal to the higher frequency in the pre-sample signal the higher the levels the less distorted a recovered signal


Advanced Television Systems Committee
ATSC is the HD standard supporting aspect ratio of 16:9
Resolutions 1920 x 1080, six times the display resolution of NTSC
ATSC defines 18 standards
Supports theater quality audio

this next section will touch on the important digital video standards to be aware of when working in cable

the ATSC or advanced television systems committee is a digital television standard which replace the analog NTSC television standard as operators moved all digital is developed by the advanced television systems committee the high definition television standards defined by the aTSC produces widescreen 16 x 9 images up to 1920 x 1080 pixels in size more than 6 times the display resolution of the earlier standard ntsc however and a host of different image sizes are also supported so that up to 10 standard definition can be broadcast on a 6 MHz TV channel through aTSC aTSC also boasts theater quality audio because it uses Dolby Digital AC3 format to

provide 5.1 channel surround sound and numerous auxiliary data casting services can also be provided through aTSC

in the world of analog video is nothing more than a collection of still pictures which when represented in quick succession on television and appears to be fluid motion videos captured at 30 pictures called frames per second presented at the scene frame rate the digital works much the same way stream of zeros and ones of data are sent to a decoder which it can reassemble them to again create the 30 frames or 29.97 to be exact the differences the bits each point or pixel in the picture must be represented by these bits which happens when the initial pictures or coded movies viewed in the movie theater are only broadcasted 24 frames per second

n the mid-to-late 1990s cable operators began to deploy a digital tier of cable to compete with satellite for your precious rf spectrum space digital is defined as an on-off state also known as 0 1 decimal 100 equals 01100100 a stream of meaningful zeros and ones can be sent to a destination computer or in our case is a set-top box with a destination device able to read the language of zeros and ones and do what is requested in the transmission the language is used by cable operators is known as MPEG-2 or H262 which allows analog video and audio to be digitized sent as a transport stream and decoded back into analog audio video the of the red in the analog world only one video can be sent in a 6 MHz channel of the videos digitizing the press with MPEG-2 10 SD or more services can be sent using the same 6 MHz channel using 256 qam

digital will always have an end of run advantage over analog provided the digital tiling threshold of the digital signal has not been crossed low signal noise snr low carrier to noise cnr cross modulation and other common artifacts will be noticeable on analog service here on the right with digital equipment will still produce the same picture that is found at headend

CEA 708 is a standard for closed captioning for aTSC digital television

carriage of vertical blanking interval VBI data in North America digital television bit streams falls under 127 SCTE BVI data transfer through the SCTE 127 mechanism is intended to be transcoded into the VBI of a companion video channel within an MPEG-2 transport stream

this ATSC a99 standard describes how to transport legacy TV data services that are sent by some broadcasters these TV data services may be sent over NTSC using lines in or near the vertical blanking interval or in digital video using VANC vertical ancillary the method for encapsulating some the services have been defined in SCT 127

digital video signals have 3 components the luminance known is y the amount of light admitted or brightness from a particular area a color value consisting of the luminance deducted from the color red or red minus Luma and a color value consisting of the luminance deducted from the color blue or blue blue minus Luma during the digitization process the 3 parameters of the component video signal are assigned a numeric sampling value

now we have 2 types of analog and digital signals here we have y PB PR and which is analog in the y PB PR describes analog component video which is like DVD players and is similar to y R minus Y and B- Y white PB PR is the analog counterpart of YC BCR white PB PR signals are YC BCR signals prior to scaling and offset 2 places signals in a digital form and are created from the

corresponding gamma adjusted RGB red green blue source it uses 3 cables for connection and is common interface for HD television sets Y CBCR is digital one of the 2 primary color spaces used to represent digital component video the other is RGB the difference between Y CB CR and RGB is that YC BCR represents colors and brightness and to color different signals which we just talked about while RGB represents colors red green and blue YC BCR the why is the brightness the CV is blue minus luminance and the and the CR is the red minus luminance a standard definition digital representation of a component analog signal set is defined in ITU RDT 601 and sent the 120 5M

a motion vector defines a region of a 16 x 16 pixel blocks in the television show the reader contains luminance samples that are used for prediction in video compression more on this topic coming up in future models the a TSC standard document a53 supports to HDTV standards 19 20 x 10 80 and the 12 80 x 7 20 that were all custom to and also allow SDTV formats including 601 format the document can be obtained at WW a TSC door and here's a nice little table shows those vertical lines of resolution the pixels that we would expect aspect ratio and picture frame rates the are different resolutions that I mentioned there was a 1080 P the 10 EDI 720 P4 80 P and the 4 EDI PB a progressive and IV and interlaced of the ITU – RDT 709 in the scent the 20 7290 6M or 4 or 280 x 7 20 P digital video was 7270 4M is for 1920 1080 P digital video standards attended repeated total pixels is 2 million as we can see here the caveat is a Teddy I is a 1920 died by 1080 image interlaced format really the total pixels are only about 1 million and then you see the total pixels for 720 P are very close to what you would see in a 10 EDI signal there is a slight example of what ST digital video and the left would look like compared easy digital video HDTV images always measure 16 units wide by 9 units high or a 16 x 9 aspect ratio dimensions were chosen to match a wide range of 35mm motion picture film standards both the 1.78:1 and the 125:1 aspect ratio programs can be shown with little or no compression the 2:35:1 or 2 to 35:1 widescreen movies can be shown with a 25% image reduction 16 x 9 was chosen because one size fits all so here's another shot of the 16 x 9 with 1.78 or 1.78:1 aspect ratio

pillar boxes where we are filling in information on the right and left insides were taken 4 by 3 and and try to fill it out with 16 x 9 is to be corrected with a nonlinear stretch but this one is the you show you the term pillar box what that means after a

4 x 3 you widescreen is letterboxing or we as we fill in the top is where you just try to view HD programs on a 4 x 3 television to correct a picture crop is needed however it can be corrected with a linear zoom window boxing is common when viewing letterboxed digital television programs

in this section relook at the characteristics of digital audio United States

the Dolby AC3 system is used for the transmission of audio and digital broadcast television and digital television for cable television systems DVD and Blu-ray formats also use the Dolby AC3 for most 60 Hz video systems the AC3 format produces a 5.1 channels of audio a front left a front right center surround laughs around right and a low-frequency enhancement baseband analog audio is 1st digitized at 48 kHz sample rate the sample is in the process by modified discrete cosine transform this process is our produces a frequency domain representation of the audio

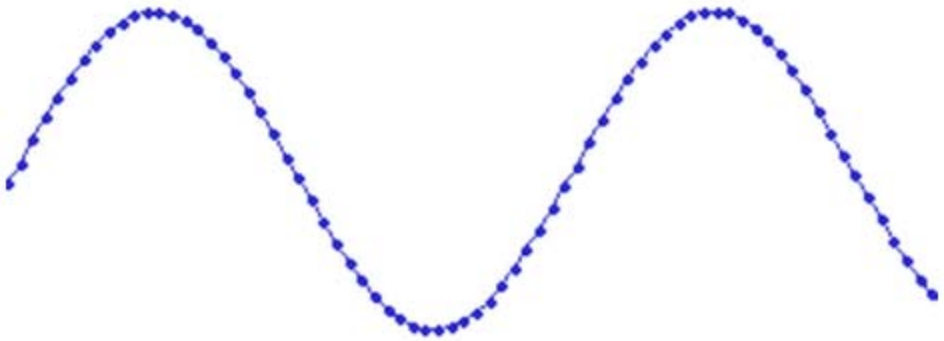
the audio is then compressed using a psychoacoustic model the optimum data rate for the AC3 compressed audio is 384 kb per second other generates are supported viewing various levels of audio quality at the receiver

sometimes an audio signal may require adjustment to correct inequality in the levels of various frequencies and audio equalizer provides a means for individual adjustment of amplitudes of the audio signal as a function of frequency typically an audio equalizer divides the audio spectrum into a series of bands that are evenly space on a logarithmic scale this is done to accommodate the human ear which hears frequency logarithmically the amplitude of each band may be individually adjusted compensating a for loss of fidelity that may occur in an audio distribution chain such as a high-end rolloff due to parasitic capacity

the transmission system for audio signals and cable television and HFC networks has a limited dynamic range that means that there is a limited range of audio amplitudes it can be sent over the network audio levels that are too low because the device is the noise resulting in a poor signal-to-noise ratio at the output of receivers on the end of the network signal that are too high may result in distortion of it all you voltage waveform is excessive excessively large the peaks of the waveform to be clipped off in the transmission is referred to as clipping distortion and is illustrated in the illustration in the lower figure on the right-hand side where you see it's clipped there between the average level of audio signal and the peak level of audio signal is known as headroom and is typically measured in DB several DB of headroom is desired to avoid distortion excessive headroom can result in poor receiver signal-to-noise ratio

audio level is measured in units of DBU 0 DBU correspondence will voltage of .774 Vrms volts root mean square with a voltage is applied to 600ohms it produces a one millivolt the nominal input level for professional audio equipment with a balance 600 ohm input impedance is around 0 DBU most audio level meters are calibrated in units of you a dbu

Congress contained in the commercial advertisement loudness mitigation act also known as the calm act requires a commercial advertisements must have audio that is average level that is no greater than the level of the audio in the programming that it occupies this requirement is reflected in the FCC regulations order levels must be monitored to maintain at the proper level this is particular important in the case of local ad insertion or commercial content may be derived from a separate source and the programming content



Using a lower sampling increases quantization leading to lower quantization noise meaning more dynamic range.

lower sampling is higher quantization we talked about sampling it came to video the Nyquist theorum right each dot in the figure above represents one audio sample the sample rate and audio CD had a sample rate of 44 kHz or 44,100 Hz often written as 44 kHz for short higher sampling rates allow a digital recording to actually record higher frequencies of sound the Nyquist rate should be at least twice the highest frequency you want to represent humans can hear frequencies above 20 kHz so 44 kHz was chosen as the rate for audio CDs just include all human frequencies sample rates of 96 and 192 kHz are become more common particularly DVD audio many people honestly can't hear the difference higher sampling sizes allow for more dynamic range louder louds and softer softs dynamic range and audio CD is theoretically about 90 DB using a lower sampling increases quantization leading to lower quantization noise meeting more dynamic range

3.

the bandwidth refers to the raw bit rates and data rates bandwidth for communication channels derived from the product of the symbol rate and the bits used in a symbol before it forla bandwidth's raw bit rate equal symbol rate times this for symbol please see the notes for clarity it was of common raw bit rates and data rates are on the screen is broken up by modulation symbol rate this for symbol raw bit rate data rate

Bandwidth

Coding schemes like trellis reduce the per bits symbol.

FEC and Encryption are overhead.

Modulation	Symbol Rate	Bits / Symbol	Raw Bit Rate	Data Rate
8-VSB	10.76 MS/s	3	32.28Mbps	19.34Mbps
QPSK	1.28MS/s	2	2.56Mbps	2.30Mbps
QPSK	2.56MS/s	2	5.12Mbps	4.60Mbps
16 QAM	1.28MS/s	4	5.12Mbps	4.50Mbps
16 QAM	2.56MS/s	4	10.24Mbps	9.0Mbps
64 QAM	5.057MS/s	6	30.34Mbps	27.0Mbps
256 QAM	5.36MS/s	8	42.88Mbps	38.88Mbps

Eight times the baud rate.

The bandwidth refers to the raw bit rates and data rates. Bandwidth for a communication channel is derived from the product of symbol rate and bits used in a symbol.
The formula for bandwidth is: Raw Bit Rate = symbol rate (baud rate) bits / symbol.

A list of common raw bit rates and data rates are on the screen, and below.

Modulation Symbol Rate Bits / Symbol Raw Bit Rate Data Rate
8-VSB 10.76 MS/s 3 32.28Mbps 19.34Mbps
QPSK 1.28MS/s 2 2.56Mbps 2.30Mbps
QPSK 2.56MS/s 2 5.12Mbps 4.60Mbps
16 QAM 1.28MS/s 4 5.12Mbps 4.50Mbps
16 QAM 2.56MS/s 4 10.24Mbps 9.0Mbps
64 QAM 5.057MS/s 6 30.34Mbps 27.0Mbps
256 QAM 5.36MS/s 8 42.88Mbps 38.88Mbps

for one example of you take a look at 64 QAM the symbol rate is 5.057 millions of symbols per second 6 bits are used per symbol we times those 2 together to get the product 30.34 bits per second for the raw bit rate however due to things like FEC forward error correction which is one example of how the raw bit rate is reduced the data rate is only 27 Mb per second other factors that may reduce the data rate are encryption and overhead requirements of the operator due to coding schemes like trellis coding less bits per symbol may be possible for instance in of VSB Vestidual sideband only 2 bits for symbol are possible instead of the 3 bits per symbol reducing the data rate to 21.52 per second overhead is also required and reduces the raw bit rate to the data rate as I mentioned for insensitivity 256 QAM the raw bit rate is 42.88 Mb per second

however the data rate is only 38.88 Mb per second another note is a QPSK bit rate is twice the baud rate 64 QAM bit rate is 4 times the baud rate and to 256 QAM bit rate is 8 times the baud rate

FEC or forward error correction is one example of how the raw bit rate is reduced other factors that may reduce the data rate are encryption and overhead requirements of the operator no one due to the coding schemes like trellis coding less bits per symbol may be possible for instance in VSB only 2 bits for symbol are possible instead of 3 bits per symbol reducing the data rate to 21.52 Mb per second furthermore in 256 QAM the raw bit rate is 42.88 Mb per second however the data rate is 38.88 Mb per second due to forward error correction note 2 QPSK bit rate is twice the baud rate 64 QAM bit rate is 4 times the baud rate into buddy 6 QAM bit rate is 8 times the baud rate

cable digital video signals require very high data rate to transport video signals over the cable network digital video signals are spatiotemporal signals having both spatial and temporal qualities in other words the digital video is a sequence of time varying images that simulate motion to reduce the high data rate a video operators use compression techniques to significantly reduce video bandwidth and the size of the video transmission over the cable network 4 types of redundancies and video are used to compress video they include 1 horizontal resolution to vertical resolution 3 spatial information 4 temporal information sampling of the horizontal and vertical resolution is used to reduce redundancies and video and in MPEG compression mpeg 4 must take advantage of the spatial and temporal redundancies between frames to achieve higher compression ratio removal of spatial redundancy occurs on a single MPEG frame and temporal redundancy is across multiple MPEG frames

Cable digital video signals require a very high data rate to transport video signals over the cable network. Digital video signals are spatiotemporal signals, having both spatial and temporal qualities. In other words, digital video is a sequence of time varying images that simulate motion. To reduce the high data rate of video, operators use compression techniques to significantly reduce video bandwidth and the size of the video transmission over the cable network. Four (4) types of redundancies in video are used to compress video, they include:

1. Horizontal Resolution
2. Vertical Resolution
3. Spatial or intra-frame coding
4. Temporal or inter-frame coding

Sampling of the horizontal and vertical resolution is used to reduce redundancies in video and in MPEG compression.

MPEG formats take advantage of spatial and temporal redundancies between frames to achieve higher compression ratios. Removal of spatial redundancy occurs on a single MPEG frame while temporal redundancy involves multiple MPEG frames or group of pictures (GoP).

another option to increase channel data rate is increasing the channel width typically return path options are 1.6 3.2 and 6.4 MHz increasing the channel width or using a spectrally more efficient modulation increases the required signal-to-noise ratio additional bandwidth is added by provisioning additional high-speed data RF channel on a broadband cable system increasing capacity when properly implemented Bonnie forward path 6 MHz channels together or return path 6.4 MHz channels offers additional bandwidth for example for bonded forward path channels offers 24 MHz of bandwidth were proximally hundred 32 Mb per second mess because it's 38 Mb per second×4 channels you've completed the lesson in this lesson you learn the relationship of symbol rate bandwidth and channel size the cable system press enter when you're ready to start the next lesson placing information on a carrier and cable whether it is data or video is called modulation modulation of information is required to transport signals over the Ford and return path communication channels for example the 6 MHz in the forward path 3.2 MHz or 6.4 MHz and return path this is to the customer the choice of modulation techniques for operators is one that optimizes bandwidth that is it improves the data rate the megabits per second versus the bandwidth used in on the operator spectrum and offers resiliency to noise in other words decreases the cost per bid transmitted keeping up with the exponential bandwidth demand growth that we see in cable there are many choices for modulating signals and this section will explore the types of modulation upon completion of this lesson you'll be able to recognize the analog and digital modulation orders used in the cable access network a carrier signal or continuous wave CW contains no information of its own provides energy at a specified

stable frequency to transport both analog and digital signals this ensures reliable signal transmission the process of adding intelligence or information from a baseband signal to the carrier by change the carrier signal's frequency amplitude phase or some combination of these characteristics is called modulation the frequency band surrounding each carrier is also known as the transmission bandwidth the carrier of modulation must remain within the allocated bandwidth there to keep from being corrupted or causing interference to other signals so for example the carrier bandwidth of the forward path is 6 MHz using modulation to transmit information falls under 4 main steps the 1st step operator will do is generated pure RF carrier at the head and transmitter at a particular frequency the RF carrier can be a sign we even analog or square wave and digital shown here the pure RF signal is modulated with the information it could be video data voice or some other type of data and that's to be transmitted to the customer the change in a signals characteristics allows it to carry this information examples of how signals are changed are by amplitude frequency phase or combination such as able to do in phase the model and in step 3 the modulated signals combined with other signals of the head and and fed into a laser transmitter and file except for a receiver in the field such as a set-top box or cable modem I will take the modulated signal detected and demodulated to recover the information there are 2 categories of modulation available analog and digital analog modulation techniques are not very common in cable but still may exist in odd modulation includes amplitude modulation a.m. now this technique is bearing the amplitude of the carrier based on the characteristics of the baseband signal the next analog modulation technique is frequency modulation or FM the technique of bearing the frequency of a carrier based on the characteristics of a baseband signal and a 3rd type of modulation offer analog is phase modulation or p.m. the technique of bearing the phase of a carrier based on the characteristics of the baseband signal other 6 different types of digital modulation and digital modulation allows operators to improve the data rate over the analog modulation and also decreasing the cost per bit are focusing cable is QAM however we will discuss some other options digital modulation includes the following we have amplitude shift keying or ask a is the same as a.m. but limited to a predetermined amplitude shifts we have frequency shift keying or FSK is simply FM but limited to predetermined ships and frequency as opposed to FM which allows for infinite degrees of frequency deviation we have phase shift keying or PSK is the same as PM or phase modulation but limited to a predetermined phase shifts we also have binary phase shift keying or be PSK this is the simplest form of phase shift keying and digital transmission utilizing one bid per symbol and for defined phase angles to take a look at shortly quadrature phase shift keying so more complex version of frequency shift keying or phase shift keying in digital transmission and it uses a 2 bit per symbol and finally QAM quadrature amplitude modulation 2 of the 3 properties of a continuous wave carrier can be used at the same time to pack more bits per baud and more bits per symbol increasing the bits for Hertz digital modulation includes ample to Jeff King or a.s. K is the same as a.m. but limited to predetermined able to shifts frequency shift cane or FSK is simply FM but limited to predetermined ships and frequency as opposed to FM which allows the for infinite degrees of frequency deviation phase shift keying or PSK is the same as phase modulation but limited to predetermined phase states binary phase shift keying or be PSK is a simplest form of phase shift keying in digital transmission utilizing one bid per symbol and for defined phase angles to PSK your quadrature phase shift keying is a more complex version of phase shift keying and digital transmission utilizing 2 bits per symbol sometimes called for QAM and then quadrature able to modulation or QAM 2 of the 3 properties of a carrier wave

can be used at the same time to pack more bits per baud and more as per symbol increasing the bits for Hertz a national television systems committee NTSC analog modulator is made up of 3 fundamental components the 1st one is a video baseband the second one is intermediate frequency and the 3rd is radiofrequency the NTSC demodulation process yields a composite baseband video signal while the digital modulation process yields a QAM video signal the death of modulation is measured as the percentage of the total amplitude change of the carrier or video voltage relative to the horizontal sync pulse level and this is as the signal progresses from sink tip to peak white AC ATV modulation requires 87.5% of the total range of the carrier envelope from full carrier to know carrier for proper performance the carrier modulation is too deep it will be cut off or exhibit reduced color and contrast during sink tips the carrier is 100% of the full carrier power a white portion of the picture or peak white level the carrier amplitude is reduced to 12.5% of the full carrier power and NTS see video signal uses 3 analog modulation techniques to generate a television channel we use amplitude for luminance frequency for audio and phase for chrominance signal processors are used to transfer signals from very high frequency VHF over the air or dedicated feeds coming into the head and over the years also referred to as OTA processors do the following they change the frequency of the signal converting it or down converting it they regulate the signal level using a automatic gain control feature AGC they amplify signal with respect to the video they modify audio levels with respect to the video processors may be frequency agile dedicated to a particular frequency were non-agile signals are often down converted to an intermediate frequency that we just mentioned an industry standard to be supported to or transported to other hidden devices the usual if frequency or intermediate frequency band for NTSC signal is 41 to 47 MHz all converters are used to move a signal to a desired frequency or output channel amplitude modulation is a simple robots modulation with the advantage of capturing nuances of change however analog signals are subject to interference noise and degradation affecting the amplitude of the carrier a.m. is an analog modulation technique were one changes the height NDB the voltage strength of a continuous waveform CW in this case a sine wave and we can see that on the right-hand side of the screen name is used at a particular frequency say 55.25 MHz for CEA channel to smaller carriers called sidebands the sum and difference carriers are produced during amplitude modulation as shown here on the screen the center carrier often represents video sound or color carrier and analog a 55.25 MHz RF carrier modulated with a 1 MHz baseband video carrier produces 3 amplitude modulated carriers wanted 54.25 MHz wanted 55.25 MHz and one at 56.25 MHz used by early modem types to convert digital signals to analog by transmitting large amplitude sine wave in place of a 1 a 0 or low amplitude for 0 and that is using in a modulation for transmitting data during 100% modulation the carrier at 32 DB will drop about 60 B to 2060 B and produce sidebands that are 6 TV down for the carrier is shown here on the screen the video information of it NTSC signal is modulated using empathy modulation amplitude shift keying array is K is a digital version of a.m. where the signals amplitude is limited to a set usually 2 or predetermined predetermined specific voltages and typically to his dumb because of binary typically referred to is a high and low or a 1 and 0 or sometimes even on and off having more possible amplitudes will result in more information being transmitted per change in amplitude pulse reoccurrence time or PRT describes the rate at which the digital voltage changes are allowed to occur baud indicates how fast the analog signals changing where the rate of change in the channel or bandwidth the speed of changing modulation is limited by the bandwidth which is a disadvantage of a.m. meaning that the bandwidth of an a.m. signal is sent in is 6 MHz

and the forward path able to shift keying is a digital version of amplitude modulation as a signals amplitude is limited to a set usually to for binary of predetermined specific voltages to refer to is high and low or 1 and 0 or on and off having more possible amplitudes will result in more information being transmitted per change in amplitude pulse reoccurrence time or PRT describes the rate at which the digital voltage changes are allowed to occur baud indicates how fast the analog signal is changing where the rate of change in a channel or bandwidth the speed of the changing modulation is limited by the bandwidth which is a disadvantage of amplitude modulation the digital TV or DTV standard chosen by the FCC for modulating over the year or off air signals United States is 8 symbol this digital sideband or 8 VSB modulation the 8 symbols represent 3 bits of data the bits are not transmitted like QAM and said the viscera added then transmitted in analog 8 VSB modulation offers a similar performance and spectral efficiency to 64 QAM but does not offer the quadrature component the 64 QAM uses spectral efficiency is how well a frequency spectrum is utilized such as a 6 MHz band with channel in our case within the forward path since cable uses 6 MHz channels a fully modulated VSB signal with both lower and upper sidebands we can see here does not fit in a 6 MHz channel because the lower sideband is for megahertz the upper sideband is also for megahertz and could see the kid sound carrier, sits out there to be almost a total of 8 1/2 MHz white that's a fully modulated VSB signal 8 VSB is used to transmit digital advanced television systems committee a TSC video signal such as your standard death digital signals and an most common the HDR high-definition video signals and pay to is also utilized to help lower the bandwidth utilized by the signals MPEG-2 is the video compression or patronization format used in over the air the list go see how VSP fits into a 6 MHz channel to conserve bandwidth and increase spectral efficiency single sideband SSB would be desired for over the year as removing part of the VSB signal however due to the nature of video signals and cable and the quality that we require SS B is not an option 8 VSB uses a full upper sideband and is a suppressed lower sideband as shown on the right-hand side notice how that differs from the SSB signal in the left-hand side that does not contain a lower sideband and doing this allows us to transmit the signal in a 6 MHz space of VSB filter is used to remove all but the vestige of the lower sideband QAM on the other hand transmits a signal using the full upper sideband and a full lower sideband 8 VSB transmits data using a single phase on the eye channel access only and not on the Q channel quadrature axis for this reason there is no reason in 8 VSB to separate the Í channels like QAM at the head and the single phase is modulated and filtered using the 8 VSB filter 8 VSB modulation is also able to transmit twice the symbol right in the same bandwidth is 64 QAM says a lower sideband is suppressed frequency modulation FM is another analog modulation technique that is less susceptible to environmental noise bursts than a.m. case in point is the radio that we have in our car FM signals a little bit improve quality FM is measured in cycles per second or CPS or frequency of sine wave CPS is the rate of change that the aura signal undergoes per second of a given wave also referred to as baud rate like a.m. the disadvantage of frequency modulation is at the rate of frequency changes are limited by the bandwidth the audio information of it NTSC signal is modulated using FM the digital version of FM is frequency shift keying or FSK FSK conveys data by periodically changing the frequency of the carrier to represent a transmitted 0 or one state ones and zeros are represented by a switching frequency for example in digital FSK and obsolete 300 Bd modem used to used thousand 70 Hz for a binary 0 while 1270 Hz meant a binary one is a little 54 United States the FM broadcast band is 87.7 MHz to 108 MHz and that overlays are cable spectrum and analog phase modulation the phase of a uniform frequency and amplitude sine wave is a specific time

intervals modulating the carrier the many phases or angles and I saw slopes of the site way can be measured and given reference values based modulation represents the on and off the ones and zeros by using the different angle shown here variations of the phase modulation allow coding of multiple bits to her for Betz on a single carrier change patient came is a digital version of phase modulation analog carrier may switch between 2 phases 19 to 70° to represents ones and zeros binary phase shift keying and quadrature Bishop King are forms of phase shift keying the color information of an NTSC signal is modulated using phase modulation here we have an example of an NTSC signal you may pause a training and take a look in more detail binary phase shift keying or BPS K is a simplest form of phase shift keying in digital transmission referred to the BPS K diagram here on the screen 2 phases are used separated by 180° and in phase and out of phase or reversed the amplitude remains constant for both in phase and out of phase signals there is a 1 bit per symbol in the BPS K diagram therefore the bit rate and BPS K is one times a symbol rate per second BPS K's extremely robust and operates well in a noisy upstream offers a low data rate and much lower than QPS K quadrature phase shift keying and quadrature amplitude modulation or QAM a more advanced fee shift keying or PSK is multi bit modulation quadrature phase shift keying or QPS K QPS K employees periodically shift in the phase of 2 in phase carriers I enqueue at a 600 Bd rate or 90° shift plus and encoding technique QPS K calls for fortifying phase changes or angles to represent data symbols 90° of separation between phase angles 0 9180 and 270° are used once a shipper. Of the wave means that each. Can represents 2 binary data bits per symbol to the barn area only represents a 0 or 1 it is based to a bit is the smallest unit of data communicated by a digital device the bits per symbol in QPS K or 2 bits per symbol in QPS K to race of the power of 2 equals 4 possible symbols QPS K's equivalent of a 4 QAM is there only to bits squared or for or 4 points sometimes referred to as symbols most often referred to as symbols but sometimes referred to as data points on on the each axis is shown here the civil bit value shown in the QPS K consolation estate diagram are 00 and he upper right quadrant 01 in the upper left 10 in the lower right and then 11 in the lower left quadrant QPS K is typically used in the upstream by cable operators for slower data services such as an out of band signaling to set-top boxes or OOB is also used to receive satellite signals in the head and or hub site the signal-to-noise ratio of QPS K should be 25 DB or greater QPS K operates well in a noisy upstream but offers a lower lower data rate and QAM inquiries travel to modulation or QAM to 3 properties of the carrier can be used at the same time to pack more bits per bought Palm also called phase amplitude digital modulation is a method for encoding digital data on an analog signal or carrier wave both phase shift keying and amplitude shift keying or varied to transmit information using QAM QAM is modulated scheme that is most suitable when bandwidth efficiency and high data rates are desired in the cable network QAM allows 2 channels and I in a queue to be carried at essentially the same frequency effectively doubling the bandwidth or data rate that can be carried while modulates to carries it at exactly the same frequency and is shifted by 90° in the Psalms the carriers back together increasing the bandwidth of forward or return path channel is achieved by using a higher order modulation such as a QAM over a QPS K increasing the symbol rate or bought using higher order modulation technique such as QAM you can Ashley use them higher symbol rates and of obviously adding symbol states moving into a 2010 24 QAM or 4096 QAM Moore symbols more bits as he order modulation increases something you need to be a need to pay attention to a higher carrier to noise ratio must exist a higher signal-to-noise ratio of at least 30 DB or greater and modulation error rate or error ratio must be maintained to offer the same bidder ratio or BER carrier noise

must also or may also be referred to as C&R carried a noise QAM is the most efficient modulation technique and therefore is often used in cable transmissions in digital modulation schemes such as QAM the data bits are sent as a consolation point called symbols or states these points represent a specific value in the in phase in 90° phase and quadrature components of the QAM signal any Q is 90° out of phase from I and the I enqueue represent phase and amplitude on the consolation higher-order QAM signals have more consolation point symbols and states that lower order QAM signals and sauce transmit more information per unit more bits per hertz Inc. proves spectral efficiency and this is in the same time the same RF channel bandwidth the number of symbols transmitted is proportional to the number of possible I enqueue values as cable operators increase modulation orders the consolation becomes denser is he really really good example of this on the screen were is less dense in the 64 QAM not sending as many symbols as the more dense 256 QAM on the right-hand side to the consolation on the right-hand side is denser the distance between the states is smaller and more susceptible to noise and transmission path distortions an increase in states were symbols in a given period of time will exhibit higher average of undetected power levels note consolation diagram may be referred to as a vector diagram digital communications modulation is often expressed in terms of the in phase I in the quadrature components a polar diagram on the left shows a symbols placement is a magnitude in the phase the polar diagram shows the I access and the 0° phase reference and the Q access is rotated by 90° there is a 90° phase shift between the I enqueue carriers on the right is the rectangular representation of the polar diagram the signals vector projection onto the I axis or I value is its eye component in the projection on the Q axis or Q value is its Q component these are the 4 quadrants of a constellation the consolation is a great tool see the health of the QAM channel your 4 examples of an unhealthy QAM number 1 in the quadrant one example a continuous wave is interfering with QAM and the symbols becoming a circle in quadrant 2 here's an example of phase noise causing the symbol to rotate in quadrant 3 example we have random noise causing phase and amplitude disturbance in the symbol is starting to spread in quadrant for example we have see reflections or ghosting are causing the symbol to replicate here we see the points in a QAM consolation for several symbols that were transmitted corresponding to that particular point in the consolation note that in the presence of noise and distortion each subsequent symbol that was transmitted falls in a slightly different location in the constellation if the magnitude of the noise and distortion is sufficient sufficiently large the received signal will fall outside the boundary the demodulator while potted incorrect value for the digital data when this happens such an incorrect output is referred to as a bit error for a given level of noise and distortion the likelihood of a bidder increases as the order of my modulation is increased or the order of QAM equipment way higher-order QAM requires less distortion and noise than lower order QAM the test equipment that is designed to display QAM consolations typically displays multiple symbols to show here in the figure how closely these points are grouped around the ideal location for a symbol is a measure of the magnitude of noise and distortion in the system modulation error ratio or myrrh is a direct measurement of the magnitude of the error or the average error of the signal from the ideal consolation location one measure of quality of a channel carrying digital data is the percentage of error-free seconds or EFS a bit at the output of the system that is different from the bit that was transmitted is referred to as a bit error one or more better is in a second constitutes 10second no errors in the second constitutes an error-free second ratio of the number of error free second so the total number of seconds measured times 100% is the error-free second percentage it is a it is

desirable to make this quantity is close to 100% as possible to 56 QAM has 8 bits per symbol in the constellation in this diagram we start on the left with a 90° phase shift between the I enqueue carriers that means that 8 bits per symbol is split between the I and Q carrier forbits each the I enqueue carriers are modulated using amplitude I is modulated to raise the power of 4 times or 16 levels of ample amplification the same is done for the Q carrier the data is transmitted over the network in our case the HFC the demodulator at the other and the text of 90° phase shift recovers the I enqueue carriers the demodulator will then detect the various levels of amplitude using the data from phase and the amplitude the consolation is created note that the NTSC demodulation process yields a composite baseband video signal while the D the digital demodulation process yields a QAM video signal here's a difference from looking at a QAM on a spectrum analyzer on the left-hand side versus looking a QAM a consolation diagram or state diagram QAM a spectrum analyzer sometimes referred to as the haystack remember QAM is an analog carrier created by able to shift keying and phase shift keying used to represent digital content on the access network here we see a QAM signal as it would appear a spectrum_haystack the horizontal access shows frequency and vertical access shows amplitude many different RF modulation schemes can coexist or on a collective network or agency network as we talked about 4 of AM is used for analog video channels FMA be used for broadcast FM band to be carried over network motor set-top boxes use BPS K for the mitigation but most dental Q PSK and an older 12 telephony services may use him to be a skater most everything's been transitioning to QAM for data services voice services and digital television it was a modulation scheme depends on the type of signal being transmitted and the cost of the RF bandwidth required 16 QAM's were symbols are equal to 4 bits per hertz to bits on the I channel into bits on the Q channel or carrier forbits per symbol yield 0 to 15 possibilities 16 4 x 4 total states or to raise to the power of 460 QAM is typically used in the upstream or return path for data services and newer set-top boxes Dr. specifies that the operator can select an upstream modulation of Q PSK or QAM on the screen is the I enqueue consolation plot showing for level IV levels of I carriers and 4 levels of Q carriers to increase the data rate and operator can use higher order modulation techniques such as 64 QAM shown here to see for QAM's were symbols are equal to 6 bits per hertz 3 bits on the I channel 3 bits on the Q channel 6 bits per symbol yield 0 to 63 plot states or possibilities that's an 8 x 8 or 64 total states or to raise to the power of a 64 QAM is typically used in the upstream or return path for Dick data services and newer set-top boxes similarly 64 was selected for the downstream or forward path Nativity says QAM is more common dots or specifies an operator can select an upstream modulation of Q PSK or QAM here's a comparison of the spectral efficiency of a VSP and 64 QAM as you can see there very similar nature of the raw bit rate being 32.28 Mb per second grade BSB 30.34 Mb per second 154 QAM the signal-to-noise ratios are practically equivalent to continue to increase the data rate over safety for QAM and operator can choose to 36 QAM activity 6 QAM requires at least the least amount of RF bandwidth for a given amount of data duties is QAM's were symbols are equal to 8 bits per hertz for bits on the I channel and forbits on the Q channel or carrier 8 bits per symbol yield 02 55 possibilities or states 236 is a 16 x 16 grid or total states or to raise the power 16 256 to 86 QAM is typically use the downstream or forward path for litter video data services and operator can select different QAM downstream channels and modulation orders for the forward path 1024 QAM has been shown feasible in a clean spectrum presidency were just going to keep increasing QAM to to increase the bandwidth to the customer but I can see help dance 210 24 QAM is compared to 36 QAM to you go denser in a clean spectrum of the 4096

QAM has been shown feasible in Europe using digital video broadcast over cable part 2 or DVC 2 for example and this represents a 50% increase in capacity over 286 QAM a signal-to-noise ratio of 46 DB or greater is required to achieve 4096 QAM here we see a summary of the QAM consolations used by operators today feel free to pause the training to to review the most effective way to analyze our signal transport or singles or transport digital data for example Susie for QAM in the upstream is a consolation analysis we do spend talking about consolation analysis is the measurement of the QAM transmission possibly measured at the customer had invocation the consolation represented good quality signal should result in good modulation error ratio and good bit error ratio good Murray Burr corresponded good data throughput characteristics few or imperceptible impairments square display points centered within the decision boundaries cluster of nearly identical points consolation analysis starts with the signal level meter in SLM with a built in QAM demodulator the QAM consolation analysis of signal level meter typically provides Burr murder RF power levels in the consolation diagram the diagram of the consolation will indicate not only if impairments exist on the single but often will enable broadband premises professional to determine the root cause of the impairment we can see on the left-hand side we have a near ideal consolation in in the middle we can see those points earned spreads with the moderate and honestly on the right-hand side to really spreading and going outside a decision boundaries to notice becomes an unacceptable to her to see poor murder due to the noise that's evidence here by the receipt QAM consolation diagram here we see the decision boundaries for several QAM types the upper left quadrant of this consolation shows a decision boundaries for 16 QAM anytime the signal falls inside the area indicated and blow it will be demodulated as a digital word corresponding to that part of the consolation noise or distortion in the signal may cause the exact value of the signal to move around inside a decision area but that is a correctable error but so long as a sin remains inside a blue area will be able to properly demodulated the lower left quadrant of consolation shows decision boundary for 16 or 6 report 1 to the decision boundary for 64 QAM is one 4th the size of the 16 QAM physical must fall within a smaller area in the in order to be demodulated properly this means a 64 QAM is much less tolerant of noise and distortion 16 QAM 086.duty 6 QAM is even less tolerant of noise and distortion because of the smaller decision boundary and that is compared to 64 QAM as indicated by the small decision boundary in the lower right quadrant of if if we have symbols landing outside the decision boundaries these are called uncorrectable's or your post FEC demodulation is the reverse process of modulation extracting the data bearing signal from the modulated carrier wave for example give operators use a demodulator notices set-top box the set-top box removes the QAM information recover the video signals let's talk a little bit about the demodulation process 1st the carrier frequency is recovered so we called carrier lock that we have symbol clock recovery which is symbol lock is step 3 seal the composition to I enqueue components of the QAM we determine the IQ values for each symbol which is called slicing that we go to decoding of the signal and the interweaving that we expand to the original Bitstream and finally digital to analog conversion of his required but the symbol clock frequency and phaser timing must be correct in order for the receiver to demodulated bid successfully and recover the transmitted information the error detection process usually begins by 1st modulating that the digital signal carrier frequency recovery symbol clock recovery signal decomposition determining I enqueue values decoding and DNA living can impact the error detection correction of the data stream you have completed the lesson in this lesson you learn the analog and digital modulation orders used in the cable access network press enter to start

the next lesson so editions of 236 QAM or other versions of calm used trellis code modulation to improve signal-to-noise like the idea of Ford error correction was take a look upon completion of this lesson you'll be able to recognize cell trellis code modulation is used to improve RF signal quality trellis coding is a technique of encoding data such that the errors that occur in the transmission channel can be corrected at the receiver extra bits are added in order to permit error detection and correction trellis coding is one of a group of codes known as Kong delusional codes as with most digital communication systems multiple bits are grouped together in the symbols, delusional codes are created by performing mathematical operations on this is on the symbols information that is sent over the channel relates to the transition between symbols rather than the symbols themselves consider a simple example of trellis coding in this example the bits are grouped into 2 bit symbols the symbols referred to as symbols ABC and D respectively as illustrated in the table or 2 to the second power gives you 4 options the bits are sequentially applied to the encoder shown on the bottom encoder consists of 3 bit digital shift register and a pair of modulo 2 adders the best from all 3 stages of the shift register are applied to the top at to reduce the I channel bits the bits in the 1st stage and last stages of the shift register are applied to the bottom at her to produce the Q channel bits note that the I enqueue channels contain information not only about the 2 bits of the current symbol but also a bit of the next symbol the transmitted information spans more than one symbol the table at the bottom illustrates how the encoder drop rates on the left or the for possible starting symbols for each starting symbol the next bid to be transmitted may be either a 1 or a 0 as shown in the second column the values in the shift register for each case are shown in the 3rd column the I enqueue output bits from the encoder are shown in the 4th and 5th columns respectively the last column shows the ending symbol the best way to illustrate the encoder operation is with the diagram the diagram illustrates the operation of the encoder over 3 symbols the horizontal axis indicates time in the vertical axis indicates the for possible symbols each subsequent symbol represents the clocking in of one additional bit for example if the initial bid is 00 or symbol a and a 0 is clocked in the 0 on the right of the symbol a it's discarded and a 0 is added to the left the new symbol is still 00 however the bid that is clocked it is a 1 and the resulting symbol is a 10 or symbol be the 2 possible outcomes when starting at 00R00 or 10 these transitions are indicated by the red arrows the rent hours are known as trajectories in this example each symbol has 2 possible trajectories next to each trajectory are the I enqueue bits that are the output from the trellis encoder the possible trajectories from other starting symbols are shown in different colors were corresponding to the starting symbol color this type of encoding is referred to as trellis coding is a diagram diagram resembles a garden trellis was not a simple example the sequence of bits to be transmitted is 001100 the initial symbol is comprised of the 1st 2 bits 00 the corresponds to NA the next bit is a 1 thoughts the next symbol is 10 or a B the trajectory from 00210 is shown in red and corresponds to an I and Q values of 11 know that is the bits corresponding to the trajectory or change in symbols that are transmitted not the symbols themselves the next bid is a one resulting in a symbol of 11 or D the trajectory from B to D is shown in blue and corresponds to an I and a Q of 01 as before it is trajectory that is transmitted not the symbol the next bid is a 0 resulting in a symbol C01 and the trajectory shown in orange the final bid is a 0 resulting in a symbol a 00 the trajectory is indicated in green is a trajectory or transition from symbol to symbol that is sent through the channel in this example that would be 110101 and 11 each trajectory contains information about the current symbol in the previous symbol note that the trajectories form a continuous path through the trellis diagram at the

receiver side a decoder looks at the trajectory bits and catalase the corresponding data bits now consider what happens in the example if there is a bit error in the second trajectory and in the received trajectories are 111101111 the 1st trajectory moves us from symbol a 00 to symbol B10 however there is no valid trajectory from symbol B corresponding to 11 there must be a better the possible valid trajectories are 10 and 01 we could test to see which is the correct trajectory if we assume that the second trajectory should have been 10 than that would lead us to symbol C01 wherever there is no valid trajectory from symbol C corresponding to the 3rd received trajectory of 01 offer if we assume that that second director directory should have been 01 that would lead us to symbol D11 the 3rd received trajectory is 01 corresponding to a valid trajectory from symbol D that must be the correct symbol in the past through the trellis the simplified 2 bit symbol example can be directly extended to symbols containing more bits and higher modulation orders you have completed the lesson in this lesson you learned how trellis code modulation is used to improve RF signal quality press enter to start the next lesson and afforward path heading 2 or more channels of different frequencies to a single piece of coaxial cable is called combining her multiplexing these channels must not affect one another as a travel over the access network to subscriber and return Path signals are combined in a different technique than the forward path using various multiplexing methods there are many choices for multiplexing signals and the section will explore the types of multiplexing used in cable upon completion of this lesson you'll be able to recognize the many choices for multiplexing RF signals the differences and features between them timeslots time division multiplexing TDM can be handled entirely by did electronics make it more popular versatile and acceptable for various uses. Devise a time on an ongoing channel into fixed number of intervals called frames within each frame or a certain number of fixed length timeslots the same timeslot in each frame is designated for specific channel each channel contributes its group of data in turn we can see that we have source a source B source E getting mucks together and there are 6 slots per cycle and the cycle we see that a has so much of the bandwidth for that cycle B and C the output is dense and sequenced in which each channel is active to equalize the transmission rate with the master clock in the TDM pulse stuffing is used we must maintain time between the source and destination for example we have each individual contributor of a constant bit rate digitized voice circuit and it has a rate of 64 kb per second is our our circuit if there are 24 of these contributors multiplexed what is the resulting output line data rate necessary to guarantee that each source will achieve 64 kb at the other end we do some simple math and you can go to into your notes that the follow along but is 60 4K times 24 lines which the 24 lines in this example representing the a 0 which equals .536 Mb per second that is a T1 or DS1 line and this 24 multiplexing 4 kb CBR voice channels we also have some overhead about 8 kb per second and are 8000 kb per second rather and is about 8 Mb and is out of T1 or DS1 making it a 1.544 Mb per second time division multiple access or TDMA used to be used by modems at this and set-top boxes to share a single radiofrequency or RF channel in the upstream from a central controller such as a CMTS VA broadband media is where all devices receive all information and broadband this data contains the time allocation called bandwidth allocation map and Oxus for CMTS and high-speed data transmission time standardization must be established in TDMA systems for to operate correctly the controlling device for high-speed data is a CMTS which provides a timing information allocation of timeslots and that's to allow the edge devices such as a cable modem or MTA to transmit a burst of data over a common media such as the HFC in our example and as none of the edge devices can receive from each other only to control and CMTS will receive we

also add guard time and is used to allow for errors in the transmission over the HFC is the upstream is bursty allocation mapping of the upstream channels required mapping of the permitted uses of the spectrum channel is assigned to specific units of many sponsor timeslots are controlled by the CMTS map message shown here map messages are sent from the CMTS to cable modems and a downstream each mini site is numbered in reference to a master timing reference which is maintained and distributed to the cable modems by the controller in our case a CMTS a given stream of many sponsor multiple slots can be assigned to a specific cable modem in response to a grant request to all camels requesting bandwidth on the contention basis or on initiative of the CMTS as an unsolicited grant maintenance ranging or some other type of transmission opportunity the map protocol data units or PDUs define the allowed usage of each mini slot through use of various types of information elements called IE's a mini slot is an integer multiple of 6.25 µs certain the -6 increments a note map messages are bandwidth in a downstream maps or map messages are sent every 2 ms would be 500 maps per second a map with a band with a 64 bytes is 32 kB per second and bent depending on the robust reports you have you could have 6 reports times 32 kB would give you not 192 kb and map messages in TDMA on each frequency is devices allocating assigned timeslots in which to transmit the upstream Mac messaging is used to configure TDMA mode in TDMA devices allocating assigned timeslots on each frequency to transmit in the upstream in FDMA discusses device for instance a CMTS is allocated a specific frequency band to transmit during the duration of the session advanced time division multiple access or a TDMA developed in data over cable service interface specification docs is 2.0 is a direct evolution of the 1.X physical layer which uses TDMA multiplexing provides a larger upstream channel of up to 6.4 MHz at 5.120 millions of symbols per second and provides higher modulation schemes such as quads trample to modulation 32 QAM 64 QAM a TDMA also provides more physical layer robustness in the form of 16 bytes of error correction upstream burst of interleaving and increased 24 tab equalizer synchronous code division multiple access or SCDMA was added in DOS is 2.0 and as a burst noise advantage over 18 EMA with noise and in channel impairments below 20 MHz a CDMA uses longer symbol times in a TDMA reducing the number of arid symbols while a TDMA is severely impacted by long-duration impulses on the upstream as CDMA by his very nature is able to survive the simple even when higher-order modulations are use up to 64 QAM as CDMA is used by cables and EMT Asa Sherry single radiofrequency channel and a TDMA's timeslot using up the 128 symbols that are transmitted simultaneously via 128 talk to or even a spreading non-interfering codes before the data is transmitted is randomly a randomized in a unique method so that each burst or dispersive data is spreadout using one of the 128 codewords by setting them on different codes that could be separated in the receiver without interference just assigning different data at this different times or different frequencies also keeps the data signals from interfering with one another by making the entire upstream usable for data transmissions as CDMA can increase the upstream capacity with no plant modifications required in the current version of a CDMA also includes the ability to give single modems a higher signal-to-noise ratio resin are per transmission so that signal-to-noise ratio per transmission without degrading the overall channel capacity this means that modems are set-top boxes in the customer premises will highly attenuated of streams will such as multi-rolling units will still operate in in the face of such upstream attenuation so this is a good thing using the a CDMA especially with noisy upstream's were showing here on this on the screen as CDMA has all the features of a TDMA plus a CDMA was added and Oxus 2.0 to deal with impulse noise below 20 MHz as CDMA is able survive long

duration of impulse noise using spreading as CDMA Framer maps many slots into code resources providing interleaving of data symbols as SCDMA uses 128 symbols in a spreader that spreads frame symbols for transmission highly attenuated upstream past may still operate using a CDMA the max channel width is 6.4 MHz max modulation rate is 5.12 millions of symptoms per second using trellis coded modulation or TCM encoded data symbols Mac messaging is used to configure as CDMA mode we can achieve much greater capacity over the age of CV the use of low density parity check LD PC error correcting coating and orthogonal frequency division multiplex and ROS DM as a multiplexing scheme and there are lots of reasons for going to OFD M 1st is channel synthesis is easier due to the granular subcarrier nature of O FTM shown on the screen second Mac layer bonding of large number of channels is much less complex with OFD M in frequency division multiplexing FTM with single carrier QAM which is what is currently used in cable downstream's sidebands of the carrier smear out the spectrum of the signal and thaw scored bands are required between carriers was DM eliminates these core bands in 2 ways 1st and OFD M the carriers are generated and demodulated a block and can be kept strictly orthogonal to each other by maintaining a precise frequency and phase relationship between each carrier peaks of the subcarrier's line up exactly with knolls of other sub carriers as shown in the figure above the results is that the carriers could be squeeze much tighter without interference would to each other so in extremely wideband waveform up to 200 MHz wide can be generated from OFD M sub carriers with each subcarrier packed up against the others and no wasted RF spectrum in between the spectrum of the OFD M waveform also drops off much faster at the edges than single carrier QAM shown here this is due to the narrowband nature of the off the M sub carriers and felt themselves and not due to the complex filtering required as her single QAM or single carrier QAM so wide middle of the M waveforms can be packed close more closely together not just quite as close closely sub carriers since the different FDNY band waveforms would generate or would it originate from separate transmitters the spectrum of OFD M waveform also drops off much faster at the edges than a single carrier QAM this is due to the narrowband nature of OFD M sub carriers themselves and not due to the complex filtering required as it is for single carrier QAM so wideband OFD M waveforms can be packed more closely together just not quite as closely as sub carriers is a different OFD M waveforms or wideband waveforms would originate from separate transmitters you have completed the lesson in this lesson you learn how to recognize the many choices for multiplexing signals the differences and features between them press enter to start the next lesson in this section will look at some of the devices used by cable operators to process video signals for transmission over the agency network upon completion of this lesson you'll be able to explain the cable operator devices used to format video signals for transmission over the HFC network here we see a single channel encoder receiving an analog signal at the head and encoder will use the analog-to-digital converter process to encode the analog baseband audio and video channel into a digital single program transport stream RSP DS and this we transmitted over the HFC or cable network to provide a digital signal at the television set as long as the individual bits can be differentiated as either a 1 or 0 a near perfect reproduction of the original information can be regenerated at the subscriber here we see a 6 channel decoder receiving 6 channels in MPEG-2 and PTS which is a multi-program transport stream is coming from the router in the head and shown here what the decoder does it converts the MPEG and PTS to baseband audio and video and this is to transmit over the HFC and were transmitting analog signals to the subscriber the next device that's used to process video signals for transmission over the HFC is the step marks

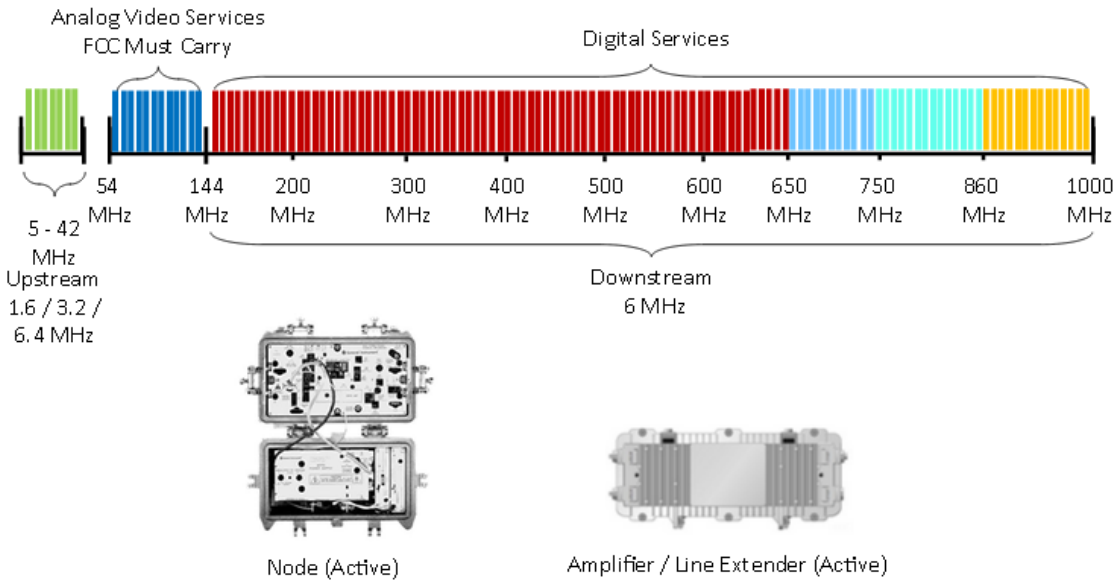
or or the statistical multiplexer and that is usage trends late real-time content into variable bit rate programs statistical multiplexing is utilized to achieve better picture quality and more efficient use of the bandwidth of the output multiplex trans rating or rate shaping is beneficial because it allows more digital programs to be squeeze into the same amount of channel space freeing up additional bandwidth for the delivery of more services operator said the claimant rate say 3.75 Mb per second for standard definition and all MPEG standard definition video streams will be shaped to this amount improving efficiency in the QAM reshaping allows non-real-time modification of encoded digital MPEG-2 video streams and other device used the process video is the edge QAM shown here on the right in red and are used to modulate off air signals or standard definition or high-definition for transport over the HFC network the edge bombs typically use a 256 QAM to modulate a 6 MHz channel for delivery to set-top boxes these are digital set-top boxes you completed the lesson in this lesson you learn how the cable operator uses devices to format video signals for transmission over the HFC

4.

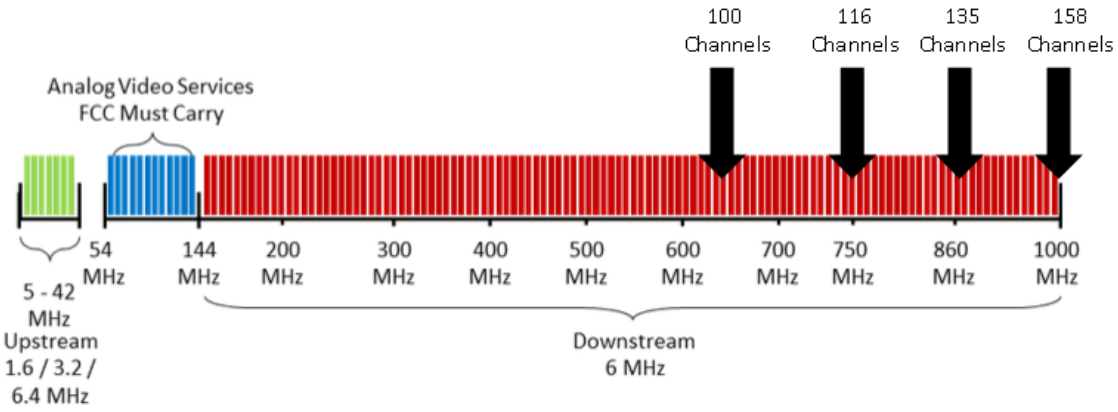
Internet bandwidth constraints and management module to start 1st in the outside plant bandwidth section in this 1st section the learner will understand how outside plant bandwidth is determined for particular architecture then take a look at the size of the RF channels in both the spectrum and the data rate in the outside plant

the outside plant has a finite amount of RF channels in spectrum per channel in the forward path the most common or if channel sizes are 6 MHz for the North American and 8 MHz for Europe

the number of our channels the outside plant depend on the size of the spectrum offered by the forward path the architecture of the outside plant determines the quality of channels offered to the subscriber and architecture will consist of active and passive devices for instance nodes amplifiers extenders in their design for a fixed spectrum typical outside plant architectures are 650 MHz 750 MHz 860 MHz and 1 GHz or 1000 other sizes may also exist but these listed here are the common architectures



each of the architecture designs use return path approximately 50 MHz when using a sub split system the most common North America is 5 to 42 MHz under 5 MHz is a bit noisy for operators to work in other systems for example Europe may use a higher split such as 65 MHz based on the docsis 3 design specification once Docsis 3.1 becomes a reality a mid split system at 104 MHz will be desired operators must subtract return path spectrum to determine the total amount of forward path spectrum the following simple equation yields an approximate forward path spectrum the forward path spectrum equals total outside plant architecture spectrum minus the return path so we apply the equation to the following spectrums so 650 MHz becomes 600 MHz trucking 50 750 MHz become 700 650 MHz becomes a 1,000 MHz becomes 950



now that we understand how much being with is allocated to the forward path we to figure out

the total number of forward path RF channels that are sent to the cable subscriber remember each RF channels 6 MHz wide we will divide the forward path frequencies orspectrum by 6 MHz to understand the amount of RF channels offered by each architecture so a 600 MHz that we are left with out of a 650 MHz system we divide up by 6 we end up with 100 channels a 700 MHz system divided by 6 116 channels and 810 MHz system by 6 135 channels and 950 MHz system divided by 6 is 158 channels depending on size your system you get more channels these are the finite amount of RF channels offered by these architectures

that we understand how many RF channels are offered we need to determine the data rate in megabits per second that is offered per channel are 2 factors determine the data rate of RF channel for the forward path and they are the modulation type used for the channel and the format of channel is it a digital channel or analog channel the modulation type used for a forward path channel determines the number of binary zeros and ones that can be stuffed into a RF 6 MHz or 8 MHz channel there are variety of modulation types available today for people operators use this in the forward path of the outside plant however 256 QAM offers the most amount of raw data today this is why it is a popular choice for the forward path note other modulation type such as 1024 QAM and 4096 QAM may be on the horizon remember what we have learned previously the higher the modulation order the more susceptible the RF channel will become when it encounters noise this means video artifacts

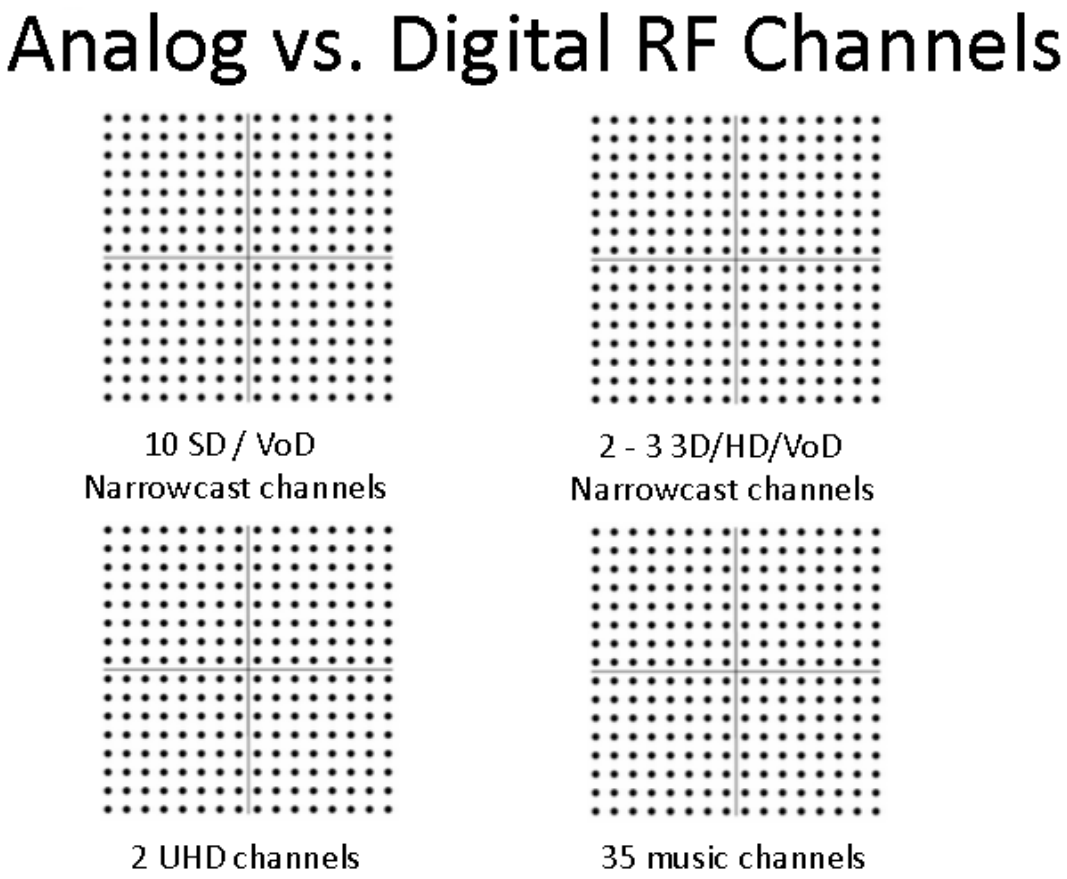
Modulation Type	Bits / Symbol	Symbol Rate	Raw Throughput
QPSK	2	1.28 MS/s	5.12 Mbps
16 QAM	4	2.56 MS/s	10.24 Mbps
64 QAM	6	5.057 MS/s	30.34 Mbps
256 QAM	8	5.36 MS/s	42.88 Mbps
1024 QAM	10	TBD	TBD
4096 QAM	12	TBD	TBD

ow what are the common raw data rates let's take a look quadrature phase shift keying or QPS K supports 2 bits per symbol in a symbol rate of 1.28 millions of symbols per second which is about a raw data rate of 5.12 Mb per second we have quadrature amplitude modulation 16 with 4 bits per symbol and a symbol rate of 2.56 millions of symbols for second which yields a 10.24 Mb per second raw data rate we have quadrature amplitude modulation 64 with 6 bits per symbol and a symbol rate of 5.057 millions of symbols for second which is about 30.34 Mb per second raw data rate we also very common one that that we just mentioned was the quadrature amplitude modulation 256 that supports 8 bits per symbol is the rate of 5.36 million symbols per second porting 42.88 Mb per second and raw data form there are a few of the future modulation types that I mentioned a quadrature amplitude 1024 with 10 bits per symbol and driving it at symbol rate is equal to 256 produces QAM human to be able to drive that in the 50 Mb per second and if you look at quadrature amplitude 4096 with 12 bits per symbol again driving that at 5.36 mil symbols will give you around 60 but the symbol rates for these I will be

much faster than than what's in 256 QAM of this little note there millions of symbols per second is MS/s

we should have a solid understanding of the amount of bits that can be produced using different modulation orders what's take a look about the second factor determines the data rate the format of the channel may be digital or analog each analog video takes an entire 6 MHz of bandwidth to deliver a piece of video to subscriber for 750 MHz architecture hundred 116 analog video channels may be offered to subscriber however subscribers want more using digital video channels operator able to multiplexer combine multiple digital video channels into a single 6 MHz channel this is why cable operators are very interested in digital signals in the outside plant in the mid-1990s digital TV started to become a reality for cable operators and for cable can customers analog signals were digitized called all digital simulcasts or ADS ADS allows more channels per QAM

ADS shown here has multiple copies of a channel perhaps one as an HD one is an SD in the maroon area and maybe one in HD in the maroon area and then one analog in the blue area the next step after ADS is analog reclamation and that's when allow you to pull back the analog of the spectrum and provide additional bandwidth



1. Standard definition (SD) has a data rate of approximately 3.5 Mbps.
2. High definition (HD) has a data rate of approximately 12 to 15 Mbps.
3. Ultra High definition (UHD) has a data rate of approximately 5 to 8 Mbps.
4. 3D has a data rate of approximately 12 to 15 Mbps.
5. Narrowcast Video on Demand (VoD) has a data rate of approximately 3.5 Mbps for SD and approximately 12 to 15 Mbps for HD.
6. Music channels have a data rate close to 1 Mbps.

Now that we understand the data rates, how much data can an outs OSP or HFC support?

digital videos offered in several formats formats drive the data rate shownthe data rates are based on forward path that uses a 256 QAM achieved by well-known compression standard called moving pictures experts group or just MPEG the most common version is MPEG-2 without MPEG-2 and HD channel would be approximately 900 Mb per second is very very important that operators are implementing some type of video compression so using MPEG here the following digital video services offered by a typical cable operator standard definition which has a data rate of approximately 3.5 Mb per second high-definition which has a Gatorade proximate 12 to 15 minutes for second something new that that is very is more of a test but will be more reality in the in the coming months is ultra high and has a data rate approximate 5 to 8 Mb per second 3-D is a data rate very close to what's in HD because of the multiple cameras have narrowcast VOD which has the data rate approximate 3.5 for SD and approximate 12 to 15 but was for second for HD

so now that we understand the data rates how many data channels can we fit in the outside plant or how much can the HFC support us to look a little map for 750 MHz architecture hundred and 16 digital channels may exist if each digital channel 6 MHz of bandwidth in a data rate of 42.88 Mb and the total outside plant bandwidth is 116×42.88 Mb per second or 4009 74.08 Mb per second is a lot of bandwidth that's at the side of the subscriber's home if we subtract overhead like forward error correction and encryption than the total usable outside bandwidth becomes 116×36 or 4176 Mavis per second or about 4.2 Gb per second what is that mean what is a 1000 one 1193 standard definition channels and 3.5 Mavis per second or 278 high-definition AC channels at 50 Mavis for second or 522 over high-definition channels at a megabit per second and not all channels in the outside plant will carry video traffic this is where the 1st bandwidth constraint is faced by cable operators that you have to share the spectrum with other services let's move into the next section of the module this section will look at options operators have to manage pain with constraints faced by MSO's not all RF channels will carry digital video information in the forward path going back to our 750 MHz plant with 116 channels that will deliver more than just literary video the plate will deliver high-speed data along with video-on-

demand and linear broadcast channels digital voice channels are usually part of the high-speed data channels and in some cases analog channels still exist is becoming more common today to offer 16 bonded RF channels of high-speed data moving to 32 qualms with docsis 3.1 is fully deployed another 8 RF QAM channels are VOD moving to 12 and 16 qualms in the future today high-speed data and VOD reduce the number of RF QAM channels by 20% if an operator is using a 650 MHz architecture then this is 24% of the total for past spectrum in some cases operators are still using a few analog channels and this will further impact the bandwidth constraint they are facing in the end it will impact the offerings to subscriber the following options exist to create more forward past spectrum we could use less VOD or maybe even use less HST high-speed data were accommodation to both use less VOD and agency qualms we could just go and update the entire outside plant to 1000 MHz we could use MPEG-4 place of MPEG-2 to compress digital video since MPEG-4 offers a better compression ratio we could also switch the linear digital broadcast channels as there are always on for the subscriber and then when subscribers tune in and out of the virtual channels will switch them but see which option of MSO's will favor the trend for subscribers to do more anytime anywhere video content viewing increase in the popularity of VOD services and IP TV competition and IP video as the pressure on cable operators to deliver increase high-speed data bandwidth with speed tears reaching 600 Mb per second using QAM channel bonding now 1000 Mb per second and beyond will soon be possible with the new docs is 3.1 physical and media access control layers.sys 31 will eliminate the bandwidth constraint of high-speed data however not initially help linear broadcast video constraints to use less VOD and high-speed data qualms is typically not an option for cable operators as this will continue to increase Sherry high-speed data qualms with several nodes called lash of design is also not a viable option to greet additional forward path bandwidth operators use load-balancing designed to split high-speed data customers between downstream qualms when traffic is increased to expand outside plant to 1000 MHz is expensive women is required lashing and cable is also required this is not a viable option for cable operators operators could use MPEG-4 in place of MPEG-2 to compress digital video to produce a better compression game MPEG-4 supports up to 100 to 1 and MPEG high-efficiency video codec HEV C supports 201 the question here would be thus a set-top box in the home support MPEG-4 or an IP TV technology these are here is that we are not quite ready for a full MPEG-4 or IP TV solution however operators are moving in this direction this is not another viable option just yet the last option is switching the linear digital broadcast channels as a subscribers tune in and tune out a virtual channels on their set-top box currently is the most cost-effective solution for operators the technologies called switch digital video and is the focus in our last section was take a look in the final section learners will discover how a technology called switch digital video where SDV may help cable operators alleviate the pressures of being with constraints in the forward path switch digital video SDV allows a cable operator create a poll of digital channels for example taking qualms that support video broadcasting her definition channels and converting them to STB QAM channels these qualms are not dedicated linear broadcast channels they allow the operator to switch channels within the QAM operators will select channels for the STB QAM based on popularity the least popular are switch channels the thinking behind this approach is that all channels will not be viewed reducing digital video consumption for SDV to gain the maximum benefit for cable operators video subscribers are set-top box tuners per node must be reduced moving closer to 250 subscribers or tuners per node a typical service group has 4 notes that would allow 1000 video subscribers or tuners per service group SDV requires return

signaling the set-top box as subscribers will tune to channel and this information is transmitted back to the operator at the hobby or head end location here we have SDV like we mentioned previously SDV requires return signaling from the set-top box each time prescribers tune to a virtual channel tuning information is transmitted back to the operator to switch the requested channel the guy data for the set-top boxes dynamically updated for other subscribers that requested the same channels that are already being viewed on the note operator typically create a qualms of STB shown here switching 80 rarely watched as the channels HD could be switch but most likely these are the popular channels the SDV design will yield qualms that can be repurposed in the outside plant for new or other services popular channels will not be good candidates for SDV and will continue to remain part of the linear broadcast lineup shown here and I provides an overview of SDV in this module we discussed the bandwidth of an outside plant

5.

Module upon completion of this module the participant will be able to demonstrate working knowledge of digital carrier measurements including the equipment and methods used such as signal level meters and handheld digital signal analyzers qam analysis spectrum analysis average power measurement modulation error ratio bit error ratio pre-panelist forward error correction code word error rate and reverse ingress in this section will look at signal level meters and handheld digital signal analyzers HDSA's are powerful networked computing devices that also include an integrated cable modem pretty modulating and analyzing cable modem signals to the currently available HDSA's with the appropriate modules may include but are not limited to the abilities to characterize RF signal level in the entire suite of traditional analog signal level meter SLM measurements described in the next section such as Han tilt scan CTP CSO and CNN spectrum analysis is usually also with Max hauled and 0 span meds RF impairments such as narrowband ingress and impulse burst noise signal leakage qualm constellation and modulation error ratio MER HDSA's also have the ability to characterize bit error ratio BER of both pre-and post forward error correction processing this may include number ratio of uncorrectable Reed Solomon code words versus the allocated words as well as aired severely arid 6 seconds Dr. speed throughput and upstream transmit power level and how most networks and devices including ethernet collects bridge functionality ethernet Wi-Fi IP premises local area networks lands premises wiring to polity fault detection and distance default devices Voit performance such as latency jitter packet loss and mean opinion score and asked digital video quality measurements and IP video performance some HDSA's also include optional probes or client devices that can be placed on every outlet in the home for certification of an entire premises network for all potential services and can generate comprehensive reports on the entire premises network and viability for advanced and triple play services signal level meter SLM has been traditionally used to measure the amount of RF signal level in the DMV at specific consumer electronics Association CEA frequencies are channels given in the CEA 540 2C standard document is typically channels 2 to 117 for 750 MHz HFC system in the appendix for more information SLM's can also measure a variety of other RF metrics that characterize the performance of the cable network such as how tilt channel scans CTP CSO and CNN however like age HDSA's modern SLM's often also include other functions such as qualm signal analysis MER pre-and post-FEC BER line voltage spectrum analysis ingress test and the return noise in phase analysis carrier wave or CW interference reflections micro reflections and return noise the most

advanced SLM's can also include an integrated cable modem hence many of the instructions below for care set up calibration and test to be performed can be applied either one of these device types as of this writing the additional measurement capabilities that differentiate HDSA's from SLM include those involving mocha ethernet LAN H PNA and certification of the entire customer premises for all advanced services delivered over cable specific HD essay and SLM operating instructions can be found in the user manual provided by the manufacturer most manufacturers have the mangled online or in the meter itself free the act calibration and checkups for the HDSA's and SLM's to measure the signal levels accurately the equipment must be calibrated and operated according to the manufacturer specifications and operating procedures many manufacturers offer regular dedicated intervals to check and verify the calibration of each units work with your supervisor to schedule regular equipment checkups ensure the battery is fully charged before you begin the day's work set of HDSA's and SLM's with a channel plan the proper channel plan for the cable system needs to be loaded on HD SAR SLM the channel plan allows the meter to look at the correct parameters on each individual channel so that it can accurately read the signals coming off the HFC network as an example the channel plan will need to be configured for CEA 540 2C channels 2 to 117 for 750 MHz HFC system or the highest downstream channel frequency used in the network channel math cable operators apply or map frequency to each channel number that only their STB's can understand therefore the channel displayed on CPE may not be the same as the channel tune to test for example the weather channel might be listed as channel 10 in a specific cable system and that's how it displays on the STB the standard frequency range for channel 10 is 100 Nanny 90 to 298 MHz however it might actually be on RF channel 23 the table above shows channels 2 to 6 from the CEA 540 2C channel plan your supervisor for the channel map for your area the following measurements can be performed with either NHTSA or and SLM consult the manufacturer's instruction manual or your cable operator for specific operations on the units you will use for each measurement RF signal power are at signal level measurement forward analog and digital RF levels other RF signal level mesh measurements Han tilt non-flatness in the cable spectrum CTP CSO measurement docs as referenced modem test as well as carrier to noise ratio and modulation error ratio MER signal-to-noise ratio SNR generalized constellation analysis thermal or Gaussian noise as well as ingress test and return bit error rate code word error rate and hem certification is important to understand how to make digital carrier measurements this section will explore this topic in order to make the following measurements on the collects portion of the network the HD SAR SLM is connected to the access or customer premises networks in one or more of the following location access network measurement locations tab output the end of the drop line prior to the grounding block premises net network measurement location input to the first-time splitter outputs of home splitters and amplifiers and wall coacts outlets in addition NHTSA with ethernet IP test functionality may be connected to other networks and the customer premises as follows the output of cable modem RJ-45 port any other RJ-45 port of ethernet-based CPE any RJ-45 wall portal connector in the premises LAN wiring finally if NHTSA supports Wi-Fi measurements the HD essay may be moved around the customer premises and Wi-Fi signal levels and performance measurements may be made if directed by your supervisor in this section will look at how Kwanzaa measured by cable operators the most effective way to analyze RF signals transporting digital data for example 256 qualm downstream docs as channel is a constellation analysis a constellation diagram is a representation of a qualm signal decision points are the dots plotted by the modem these points will fall into the boundary of a decision

area which corresponds to a sequence of zeros and ones this is called a qualm constellation constellation analysis starts with an HD essay or SLM with the built-in qualm demodulator the qualm constellation analysis of the HD SAR SLM typically provides MER power levels in a constellation diagram the diagram of the constellation will indicate not only of impairments exist on the signal but often will enable the broadband premises professional to determine the root cause of the impairments there are 3 typical types of impairments that occur in an HFC network random noise primarily due to thermal or gossip in noise or occasionally also due to phase noise coherent interference from carrier wave ingress CTP or CSO and compression intermittent the following descriptions and pictures illustrate how these impairments will be indicated on a constellation diagram figure 8 shows an ideal constellation diagram with little to no impairments of any form where each received symbol scores a bull's-eye and that it plans exactly in the box intended for since a small amount of noise interference and nonlinearity is generally acceptable figure B shows a more typical constellation diagram where received symbols don't hit the exact center of each box but fall inside the box well enough to be to not be misinterpreted as the wrong symbol finally figure C shows a channel that has too much noise interference or nonlinearity and thus the constellation points here into nearby regions which leads to excessive errors constellation diagram analysis provides the earliest view of digital transmission quality problems resulting from noise ingress and nonlinear distortion the constellation diagram is thus a measure of how the transmission transmission symbols in phaser I in quadrature or Q also sometimes called phase angle state or phase amplitude state are actually placed compared to ideal placement dots should be mostly round around each constellation point for Goss in noise the more blurry the plot of received symbols in the constellation diagram the worst that is lower the MER score is significant phase noise is also present the constellation diagram will appear as if it is rotating in a circular pattern as shown in figure D if on the other hand the Quan signal is experiencing coherent interference the dots will have a doughnut form as shown in this figure this can result from a carrier wave ingress signal that is in band or from intermodulation products such as CTP or CSO or carrier spurs if the outermost clusters of dots in the constellation diagram are moved inward toward the center of the entire diagram it indicates gain compression in either the RF amplifiers RF filters up or down converters or by equalizers in the HFC access network or home amplifiers this type of impairment is typically an indication of a problem outside the customer premises such as in the cable plant or in the head and have this problem should be reported to the supervisor as a potential issue that may require escalation to plant and had in technician finally if the impairment is intermittent say due to first-order impulse noise or from laser clipping the constellation diagram a look normal except for a number of points that are unusually distant from the rest of the cluster of constellation points as shown in this figure these impairments are found and resolved by following the troubleshooting techniques depicted in module 13 troubleshooting and repair you'll find the constellation analysis great troubleshooting tool when tracking down RF impairments in the HFC network many of the newer qualm analyzers have a unique feature called qualm arrow sector spectrum EVS that allows cable operators to see digital ingress or other interference under 64 or 256 qualm digitally modulated signal without turning the Quan carrier off qualm arrow sector spectrum EVS may be used to troubleshoot and channel ingress or noise typically in channel ingress or noise sources include laser clipping loose connectors or up converter problems at the head end in this example the qualm analyzers peak marker is auto placed on a be my -30.1 DB which is located .251 MHz above the digitally modulated signal center frequency in this section

will look at how the spectrum is measured by cable operators another digital quality measurement used to determine the health of the digital carrier is a spectrum analysis a signal level meter or handheld digital signal analyzer will offer the spectrum analysis test connect the signal level meter to an RF port at the subscriber's premises to take this reading in the forward pass return Path measurement is typically done at the head end so installer will use a reverse ingress test at the subscriber location instead the reverse ingress test is coming up shortly this section will explore average power measurement a digital quality measurement used to determine the health of the digital carrier is the average power measurement signal level meter or handheld digital signal analyzer will offer the average power measurement test connect the signal level meter to an RF port to take this reading the reading is expressed in DV MV and an ideal reading modem or set-top box is 0 DV MV this section will explore modulation error ratio or MER measurements to better understand modulation error ratio or MER we should 1st review signal-to-noise or SNR signal-to-noise ratio or SNR refers to the ratio of the D modulated signal to the demodulated noise at the output of the network MER is closely related to SNR for qualm channels in both SNR and MER higher values indicate better quality and performance in fact some instruments may refer to MER S SNR even though they are calculated quite differently in digital modulation schemes such as qualm the data bits are sent as constellation points these points represent a specific value of the in phase and quadrature components of the qualm signal higher order qualm signals have more constellation points than lower order qualm signals and thus transmit more information per unit of time in the same RF channel bandwidth MER is the average amount by which the received constellation points deviate from the ideal locations on a constellation diagram modulation error ratio MER and signal-to-noise ratio SNR modulation error ratio is the qualm equivalent of signal-to-noise ratio in both cases higher values indicate better quality and performance in fact some instruments may refer to MER SNR even though they are calculated quite differently in digital modulation schemes such as qualm the data bits are sent as constellation points these points represent a specific value of the in phase and quadrature components of the qualm signal higher order qualm signals have more constellation points than lower order qualm signals and does transmit more information per unit of time in the same RF channel bandwidth modulation error ratio is the typical amount by which the received constellation points deviate from the ideal locations on a constellation diagram constellation diagrams and test using them are described in the following slides the ideal location for the symbols in the qualm constellation shown here as the small crosses the actual received signal will deviate slightly from the ideal location the modulation error is the difference between the ideal location and the location of the received symbol the magnitude of this error when averaged over many symbols is the modulation error the ratio of the average demodulated signal power to that average error power is the modulation error ratio or MER a typical minimum MER for 256 qualm at the output to the CPE for signals that have a moderate amount of forward error correction FEC added to them an event equalized should be no less than 33 DB with an ideal reading of 36 DB or greater the actual minimum level required depends on the actual modulation order of the signal in other words 1664 or 256 qualm the amount of FEC other signal processing such as Trellis coded modulation TCM etc. but a rough guideline is that for every factor of 2 reduction in modulation order 3 DB last MER is required hence if 33 DB MER is required for 256 qualm than 30 DB would be required for hundred and 28 qualm 27 DB for 64 qualm and so on please check with your supervisor to verify recommended levels of MER this section will explore bit error ratio for measurements error ratio or per is the ratio of the

number of arid bits received to the total number of bits sent in a digital communication system is extremely important to digital communications and thus in this section a fairly comprehensive description of it will be given an ideal communication system has a burr of 0 but most practical systems have gone out of millions or even one out of billions of bits in error and does have a for of one D my negative 601E -9 respectively bit errors and likewise codeword and packet errors are determined in a variety of ways in digital communications and some bit errors are actually corrected by the receiver processing using a process known as forward error correction FEC when forward error correction is unable to correct the bit errors a group of bits called FEC codeword is an error in those bits are added to the total number of arid bits the HD essay or SLM may display total per number of corrected FEC codeword errors number of uncorrectable codeword errors packet error rate a constellation diagram of the received signal pre-and post-FEC burner or any combination of these quality quantities bit errors which may be correctable or uncorrectable in a constellation diagram can be seen by constellation points that are so far away from the ideal location that they're erroneously classified as other symbols by the receiver the top figure illustrates an example burr measurement output of an HD essay or SLM while the bottom figure shows a constellation diagram with too many uncorrectable FEC eras that type that would lead to an overall burr that is too high burr is degraded or increased by our variety of issues that can occur in the HFC network and customer location among which are excessive attenuation in the path which leads to low carrier Jeanette noise ratio and/or lower low MER in band interference from radio signals such as Hammer CB radios burster impulse interference from power supplies and motors nonlinear amplification effects in the HFC plant from CTB CSO and nonlinear connector behavior such as common path distortions CPD since low levels of bit errors can often be corrected by error correction processing and cable devices bird tests help locate intermittent problems as well as find potential problems before they are seen by the customers however if only a single error is detected in the bird test it could be a very long time or short time before the next error is seen hence to get a high competence in the measurement a 95% confidence interval about 50 eras must be detected this means that if the target burr is one he -81 error per 100 million bits transmitted and the data rate of the signal is 38 Mbps appropriate for 6 MHz 256 qualm signal then 50 E positive 8 bits must be sent over 38 Mbps channel or the signal must be tested for about 138 seconds as a result most HD essay than SLM will have the option to run a 202nd bird test for an accurate reading of the quality of the 6 parts 256 qualm channel in terms of bit error rate note that is more docs is 3.0 devices are employed and cable networks the data rate can exceed hundred megabits per 2nd via bonded channels is the data rate is almost 3 times as fast this means the same 95% confidence burr test could be performed in one 3rd of the time or about 76 seconds while only the newest HD essay or SLM devices may include the ability to measure bonded docs is 3.0 channels this capability will likely be supported in the future and all such devices the HD essay or SLM will typically allow the bird to be read as pre-burr and post burr pre-burr is the raw channel bits received in error as a ratio to the total number of bits sent and since FEC can often correct these errors it may not accurately reflect true customer experience pre-burr is expressed in scientific notation for additional information refer to the metrics prefix in the appendix at the end of this module but if there pre-significant pre-burr errors even if there all correctable by FEC a can mean a channel that is barely nominal in performance and one that warrants further troubleshooting to determine the source of the bit errors post spur is the number of bits received in error that could not be corrected by FEC in the search true transmission errors that may affect CPE

performance these are also called uncorrectable FEC codeword post burr is also expressed in scientific notation while post burr errors may still not be noticeable by customers due to Internet protocols such as TCP/IP which request retransmission of arid packets or MPEG decoders which mask errors in received video packets there nonetheless real and can equate to CPE service interruptions or outages when high enough this figure should also show the example output from an HD essay or SLM showing both pre-and post-FEC burr measurements to ensure extremely high quality signaling in cable networks the minimum RF signal measurement for burr at the input to the CPE expressed in scientific notation 1E -9 or 1 bit error in 1 billion bits past with an ideal signal measurement of 0 he -0 when these levels are not present follow the guidelines and module 13 troubleshooting and repair if you're unable to solve the problem follow your company's procedures are ratio or per measures how often symbols are pushed into neighboring symbols territories causing the symbols to be misinterpreted burr is expressed as the ratio of arid bits per some number of bits sent given us the power of 10 since the number of errors are very small compared to the number of bits transmitted the test measurement is typically expressed in scientific notation for example 1 error out of 1 million bits would be expressed as 1 over 1, 000, 000 or 1 million or 1.0 lead to the -6 power here we have a chart showing different for measurements in scientific notation let's discuss pre-and post forward error correction FEC measurements qualm communication systems include the means to patch up some of the bits that become corrupted in transit over the outside plant or OST forward error correction is included with the qualm data transmissions forward error correction or FEC if the information the qualm receiver uses to fix the misinterpreted bets because pre-and post bit error ratio burr data quality made debt different greatly bit error ratio measurements are typically specified as being either pre-bit error ratio or post bit error ratio to indicate whether the data has already been repaired by FEC on this screen is an example of a poor pre-bit error ratio measurement were 6 errors for every 1 million bits is occurring using forward error correction the plant can heal as shown in the post bit error ratio measurement the post bit error ratio measurement shows less than 1 error for every 1 billion bits sent the important take away here is that this plan is suffering from bit errors that need to be addressed even when the post forward error correction or FEC is good it's useful to know how hard the forward error correction is working to correct errors we can measure the pre-forward error correction performance in 2 ways the 1st is using a bit error ratio measurement with forward error correction turned off however this is impractical in operating cable systems or we can examine the corrected code word error rate or CW ER we can also monitor the uncorrected code word error rate is described in the docs specification moron code word error rate CW ER in our next section this section will explore codeword error rate or CW ER measurements many of the docs is 3X CPE and Oxus to X EMT A's offer an additional metric related to the ER per to measure RF quality called codeword error rate or's were as mentioned earlier this is related to the forward error correction FEC signal processing and indicates errors that may not show up in the post-FEC measurement the codeword formula is total uncorrectable codeword over total correctable codeword plus total on aired codeword plus total uncorrectable codeword using an example screenshot of codeword error rate or CW ER from a modem using Internet browser open a connection to HT TP://192.168.100.one 2nd table shows energy per symbol to noise density for 256 and 64 qualm respectively a 64 or 256 qualm signal should have a codeword error rate reading of less than or equal to 10 to the -7 this section will focus on reverse ingress measurements and ingress test in return used spectrum analyzer function of NHTSA or SL M to

scan for upstream or return Path interference that can cause excessive bit errors in the upstream signaling from the customer premises examples of upstream interference include impulse or burst noise from motors switching power supplies are even dimmer switches narrowband RF interference from Hanan CB radios HF broadcast sources such as the voice of America and nonlinear signal impairments such as common path to distortion CPD most upstream interference on the HSC network comes from the drop system and the home network as much as 90% of interference can arise from these 2 portions of the network align the use of low-tech values like 48 and 11 makes it easier for ingress and noise from the home and drop system to enter the HSC network defective fittings splitters cable jumpers wall plates and poorly shielded CPE there located next to motors or other noise sources could all contribute to the interference that eventually makes its way back onto the HSC network if not eliminated or reduced by signal processing upstream interference can cause problems with two-way services such as VOD IPPB HSI and void the figure shows an example spectrum analyzer mode output on HDSA or SLM that indicates the presence of narrowband ingress in the channel of docs is upstream signal the ingress is clearly seen when the doctor signal is not active since the vast majority of upstream interference comes from the home were drop line testing the noise at every customer premises is just as critical as ensuring proper signal power levels MER and BER scores for the health of the HFC network this is an ideal way to verify the quality of the drop system and provides a good indicator of the integrity and quality of inside wiring and thereby permitting the identification of poorly shielded or damaged cable in the customer premises for example spectrum analyzer mode of these devices the ideal tool to view return Path interference other HDSA's or SLM exist to measure carrier to noise and ingress ratio and upstream transmit power required for CPE to measure the return Path noise of a customer's drop configure the NHTSA or SLM spectrum analyzer mode to upstream specifications provided by your system typical RF spectrum settings in the US are between 5 MHz and 42 MHz take a reading of the spectrum analyzer noise floor prior to connecting NHTSA or SLM to RF add a 75 ohm termination to the end of the jumper record the peak noise floor for for example -40 5D BMV unplug from commercial power any CPE devices that are connected to the cable network this will prevent a device from transmitting and raising the noise measurement following best engineering practices terminate any unused wall ports and splitter ports or as directed by your company connect the cable from the ground block leading into the customers home to the RF input port on the NHTSA or SLM so it's displaying the interference coming from the customer's home take a peak level reading while the meter is connected to the customers inside wiring for example -30 D BMV in this example we have -30 D BMV minus -40 5D BMV or 15 DB of added interference coming from this customer's premises wiring and equipment know that this is a total interference power level measurement the interference in a more limited frequency range used by CPE devices may be much lower but this total level is indicative of the wiring integrity or presence of specific interference in the customer premises always consult with your system for specific procedures for testing return Path noise to check for intermittent ingress or impulse burst noise interference the spectrum analyzer mode can usually be adjusted to the peak hold mode to capture transit signals ingress scan displays can be saved for printing later or for uploading if unusual behavior is seen in this module we demonstrated working knowledge of digital carrier measurements including the equipment and methods used for the following signal level meters and handheld digital signal analyzers qualm analysis spectrum analysis average power measurement modulation error ratio bit error ratio pre-post forward error correction

codeword error rate and reverse ingress welcome to the digital carrier measurements module upon completion of this module the participant will be able to demonstrate working knowledge of digital carrier measurements including the equipment and methods used such as signal level meters and handheld digital signal analyzers qualm analysis spectrum analysis average power measurement modulation error ratio bit error ratio pre-post forward error correction codeword error rate and reverse ingress in this section will look at signal level meters and handheld digital signal analyzers HDSA's are powerful networked computing devices that also include an integrated cable modem for D modulating and analyzing cable modem signals to the home currently available HDSA's with the appropriate modules may include but are not limited to the abilities to characterize RF signal levels in the entire suite of traditional analog signal level meter SLM measurements described in the next section such as Hom tilt can CT be CS

6.

The introduction to MPEG compression module our speaker for the module is Stephen Harris Senior Dir. of advanced network technologies at the SCT Mr. Harris has spent over 23 years working in the networking and broadband fields and his expertise spans IP routing and switching broadband cable services wireless communications and video engineering Steve has the distinction of being the 2007 inaugural recipient of the SCT E excellence in cable telecommunications learning and development award upon completion of this module the participant will be able to describe why cable operators are interested in compressing digital video understand the basics of compression is in the oldness of encoding explore how MPEG is used in cable to compress and transport digital video understand the ATS C standard is for the differences between MPEG standards and they want MPEG to MPEG 3 MPEG-4 and MPEG H explore MPEG digital audio standards why is cable interested in compression listing to look 1st was talk about how video information is sent over the airwaves in over HFC cable access networks over the years the Federal Communications Commission or SEC have created a frequency allocation map which assign specific RF frequencies of the international telecommunication Union's ITU very high frequency VHF or 30 MHz to 300 MHz and also the ultrahigh frequency or UHF at 300 MHz to 3 GHz channels this map is nonnegotiable in all broadcasters and manufacturers must follow it to ensure compatibility one all equipment and also to avoid interference from channel to channel at different points throughout the years the federal authority for example the US's FCC has also allocated a frequency map to cable operators and multisystem operators amiss owes because this map was implemented throughout different periods and because lower frequencies are less attenuated on coaxial cable the cable map does not match the off air map for all channels and moreover the cable channel frequencies are not always in order and RF video channel is defined as 6 MHz of space in the United States and many other countries and 8 MHz in Europe it takes a full 6 or 8 MHz channel to broadcast a single analog video using amplitude modulation with synchronized audio using frequency modulation this RF generalization was set for example by the national television standards committee or NTSC in United States and phase alternating lines pick up power PAL in other countries but this approach of using analog signals for TV transmission is very inefficient if the video signal is digitized the resulting video stream can be encoded and compressed without noticeable loss in video quality and set much more efficient in the network in fact many digital video programs can now be sent over a single 6 or 8 MHz RF channel in modern cable networks so in the mid-to-late 1990s give operators began to deploy a digital video tier of cable of this was

necessary both to compete with satellite competitors who had recently launched an all digital TV transmission and more importantly to use the precious RF spectrum space more efficiently so that more video content could be sent to the subscriber digital video signaling means that video content is sent as a file or real-time stream of data for the digital video receiver must know how to decode the digital video file or stream so that requires a standard the 1st standard used to deliver digital video used by cable operators is known as moving pictures experts group or MPEG version 2 also known as H262 using analog to digital ink digitally in code with MPEG allows analog video and audio to be digitized using the MPEG-2 encoding and decoding portion of the standard and sent as moving picture experts group transmission stream or MPEG TS using the transport stream portion of the standard the MPEG-2 transport stream or MPEG TS defines how the digital data is packet sized for transmission over data networks it must adhere to MPEG-2 encoding decoding and transmission stream standards allowing the digital video receiver to decode the digital video stream and converted back into analog for playing on analog televisions were to send the decoded video stream to digital monitors and display such as laptops tablets or smartphones another benefit of digital signaling for video transport is at the video signal could be sent at different resolutions standard definition digital video or SD TV is similar to the old analog TV signals and resolution while high-definition or HDTV is much more vivid and has a much higher resolution so the picture looks much more detail to the viewer but HDTV digital video requires more bits of digital information to be transmitted so fewer HDTV signals can be packed into a 6 MHz RF channel then SD TV signals while in the analog world only one video can be sent in a 6 MHz channel using MPEG-2 10 or more standard definition services may be sent using the same 6 MHz cable channel using a modulation format notice to V6 qualm Quan stands for quadrature amplitude modulation is how the digital bits are actually sent over RF channels on the HFC access network let's see how the math works out to be 6 qualm signaling is a 6 MHz RF channel which gives a digital bit rate of 42.88 Mb per 2nd but about 4 Mb per 2nd out of the total are needed for forward error correction or FEC to ensure error free reception of the digital video so we have a digital data rate capacity of that 6 MHz RF channel of just over 38 Mb per 2nd as a single ST digital video stream initially talk about 3.5 Mb per 2nd that means 10 standard definition streams require 35 Mb of data rate to send while easily fits into a 6 MHz 76 qualm channel with over 38 Mb per 2nd of data rate capacity we need some extra capacity for video video metadata and interacted interactive signaling for example to us why 10 was a good number initially modern encoders are better than the original ones however so now ST digital video streams can be sent at the same video quality with a lower bit rate sometimes even less than 3 Mb per 2nd of the video is not very dynamic so more than 10 ST channels can be set in a modern multiplex of digital video streams but high-definition digital video takes much more capacity and HD video stream can take 12 Mb per 2nd initially although again it depends on the dynamics of video but in the case of HD also on the resolution or format of the HD video the HDTV format may be for ADI 1080 P which is the original DVDs used 720 P 10 ADI and 1080 P which is a more modern Blu-ray disc players and many HDTV channels often use and there are even more higher resolution formats that are being demonstrated such as for Cayenne and 8K ultrahigh definition or you HDTV as well as a higher dynamic range or HDR TV which also require more bits than the original HDTV signals more on formats will be coming up digital video sent over the airways to HDTV's with higher gain antennas as compared to the simple extensible antenna rods of older analog television sometimes called rabbit ears the 6 MHz digital TV broadcasters mission spectrum utilizes a vestigial sideband or VSP modulation format with 3 bits

which is 8 states per symbol to to the 3rd hence it is called 8 VSP in analogy to 16 qualm over cable it has 16 unique states per symbol 8 VSP modulation has transition regions that are 620 kHz at both ban edges so that the adjacent signals do not cause interference with each other here you see a table comparing the different bit rates possible in off air MPEG video over versus over cable clearly cable is the capacity winner even at 64 qualm but for the more typical 203 6 qualm down streams and cable about twice as many programs to be sent in a single 6 MHz channel versus what broadcasters broadcasters consent of the airwaves and just wait until DOS history one comes around with 4096 qualm and maybe even higher and while a digital off your broadcast consent up to 5 SD programs the most typical case for major programming networks like ABC CBS NBC artisan 1 HD version of their main channel events and ST versions of along with 1 to 2 additional SD programming channels such as a vintage TV programming analog to digital or analog to digital converter is a process used to convert analog video to digital video equipment use is known as a coder decoder or more commonly referred to as an encoder decoder the encoder converts analog signals the digital binary bits the decoder converts digital binary to its analog signals or sine waves a Kodak performs 4 steps during the analog-to-digital process the 1st 3 steps of analog-to-digital conversion or sampling quantization and encoding these steps determine the quality of the digitized signal transmission abysses fixed with Eagle gaps and is known as iso-chronic transmission we cannot have uneven gas between ethernet frames used to transmit MPEG frames in the cable network in the final step of the analog-to-digital conversion is known as decoding in interlaced scanning one of the 2 scanning methods used by televisions interlaced being the 1st and progressively in the 2nd and images produced by illuminating individual horizontal lines beginning with lines 135 etc. and then illuminating 245 etc. standard analog television sets utilize 480 visible horizontal lines to create a single frame of video high-definition digital television sets may utilize a 1080 interlaced horizontal lines known as 10 ADI HBO and Showtime at one time use the 10 ADI format and have now switched over to a 1080 P format a digital display illuminates individual lines of pixels starting from the top left line by line and moving to the right bottom of the screen on number lines are reproduced 1st as I mentioned then displays then the display returned to the top to do the even number lines these 2 half images are interlaced to create the complete image the 2nd method of scanning is called progressive scanning and is considered superior to the interlaced scanning method progressive scanning a feature of some advanced television formats produces the frame sequentially for example lines 1234 etc. 720 P format is an example of this 720 wines are produced sequentially to create a full frame 720 P is used by many of our top programmers such as ABC WB network Fox ESPN because of the smoother image is desirable for fast action sports and telecasts it should be noted that some television sets have a preferred format for his sins of the preferred format is 720 P the set will convert any received 10 ADI signal to this 720 P format some the different resolutions you'll find under the progressive are 480 P7 20 P there's 3 different for different flavor spartan you could check this out and notes in a 1080 P format at 19 20 x 10 80 also expect to see some progressive scanning methods as we could start to see some of the 4K sets at the market the horizontal scan rate or horizontal frequency is the number of horizontal lines of information a video display can draw on the screen or frame in one 2nd and its value is given in hertz it could also be known as sweeping from the left side to the right side one horizontal line of the time a standard definition horizontal frequency is 15.734 kHz which is for 480 interlaced SDTV 31.470 kHz is used for the progressive SDTV or 480 P 45 kHz is the horizontal frequency for HDTV that uses 720 progressively scan lines or 720 P and 33.750 kHz is

used for HDTV at 10 ADI horizontal scan rate or frequency is always higher than the vertical frequency since there are many lines per frame in the video image vertical scan rate vertical frequency or frame refresh rate refers to the number frames a video display can draw or display in one 2nd the standard refresh rate for NTSC video is 60 Hz the display changes the entire video frame 60 times per 2nd vertical scan can also be known as sweeping from the bottom to the top of the screen 1 vertical line of the time in nonprogressive video the vertical frequency 60 Hz is about the same as the frame rate 59.94 frames per 2nd in interlaced video the vertical frequency usually is one half the field rate or for the 9.94÷2 which is 29.97 frames per 2nd to reduce the blurring effect that can occur on LED TVs the refresh rate was double to 120 Hz and then again to 240 Hz HDTV images always measure 16 units wide by 9 units high were 16 x 9 dimensions chosen to match wide range of 35mm motion picture films to both 1.78 x 1 and 1.85 x 1 programs can be shown with little or no loss of video information or reduction in screen size while the 2.35 x 1 widescreen programs can be shown if the video images shrunk by about 25% some 16 x 9 SD content now is down converted to 4.3 using a letterbox format as shown here letterboxing is unavoidable when viewing the entire image of HD programs on a 4 x 3 television however the zoom feature on some set-top boxes can offer a better viewing experience here is 16 x 9 polar boxing some 4 x 3 SD content now is up converted to HD using pillar boxing bars bars maybe black or gray per boxing is used when viewing 4 x 3 programs on a 16 x 9 television although some HDTV's including reformatting method were by the 4 x 3 images stretch horizontally into the 16 x 9 format by stretching the outer edges more than the inner portion of the screen and cropping some of the image at the top and the bottom this still distorts the image to some extent but to be less noticeable than the black or gray bars and a plasma organ organic LED TV says this is preferred so that the black or gray bar edges do not get burned into the plasma or oh LED screen compression is used for audio video and data bit rate reduction in cable compression is is basically the center of digital video and is used to deliver SD HD 3-D and now even the older HD to set-top boxes and clear Quam VOD content is stored and delivered to the subscriber in a compressed format using MPEG over-the-top video I mentioned earlier uses an MPEG-4 version to compress and combines it with a technology called adaptive bit rate with the size of video files compression today is required and operators will continue to look for ways to optimize video content storage and delivery let's take a look at compression let's see what in standard definition channel would look like without compression so the uncompressed data rate for progressive is the color depth times of vertical resolution times of horizontal resolution times of frames per 2nd if are doing interlaced we could take that same formula divided by 2 and the could see that one in the middle of the slider so for example we had SD video with 4 ADI with a 7 20 x 4 80 resolution 8 bit color depth at 29.97 frames per 2nd if we compute that formula we get about 41 Mb per 2nd to think about that how would you transfer 41 Mb per 2nd over single Quam you just can't do it so that means that one SD channel would take over a single 236 Quam but using compression and by by today's cable operators we can get NST down to about 3.5 Mb per 2nd here's another example of HD as same types of formulas that we saw the last one but must look at the progressive formula this time in our example we have HD video with 1080 P progressive is a 19 20 x 10 80 resolution 8 bit color depth and 59.94 frames per 2nd at uncompressed data rate if we work out the formula comes to 994 Mb per 2nd again how many Quam stewardesses qualms would that be so you can see that these are very large numbers and if we didn't have compression we would not be able to send HD NST over to our subscribers now typical as HD for today we can do bike by using the cable operator compression methods we can

do about 15 Mb per 2nd that's on average depending if you're using 1080 P or 720 P having a cell is the need for reducing the bit rate of digital television through compression several different commercial formats can be reduced using compression we look at the uncompressed HD high definition which was 1000 Mb per 2nd approximately we also looked at the uncompressed HD standard definition and we saw that floating around 40 Mb per 2nd are currently the MPEG-2 is the dominant format used in commercial digital television services but MPEG-4 is another format which is also widely deployed and there's a new one ATV CHD which is also very very popular to motion JPEG compresses the frame of the television frame independently each resulting compression ratio is about 3 to 1 for CCI R601 digital frames not very popular is only 3 to 1 how we also have DV 25 is used in many consumer cam camcorders and offer slightly better compression with each frame still in the penalty compress the resulting bit rates about 25 Mb per 2nd for HD MPEG-1 MPEG formats take advantage of spatial within the frame and temporal cross the many frames and the idea is to achieve higher compression ratios and it does MPEG 1 also usable is uses a lower pixel resolution and the bad thing about it is discards three quarters of the pixels before compressing the remaining pixels and at that's called a lossy technique and it's real lossy MPEG-2 is a dominant firm for digital video compression for SD and HD and takes Vantage of the interframe redundancy to achieve compression ratios of more than 50 to 1 but maintains a visual quality of the original unlike MPEG-1 who throws out a bunch of and then we have the MPEG-4 advanced video codec ABC part 10 is also VC one video codec one not very popular and other advanced codecs are new formats which include more sophisticated tools and MPEG-2 to achieve a higher compression ratio without losing image quality they are popular for over-the-top video in IP television commercial deployments MPEG-4 AVC offers 100 to 1 and maintains the visual quality of the ritual and I also mentioned that MPEG HE VC of the high-efficiency video codec which is the newest compression standard for reducing the size of video streams it offers close to twice a savings MPEG-4 MPEG HEV C offers close to 200 to 1 and maintains a visual quality of the original last but not least the video codec is a device or software that enables video compression or decompression for digital video one very significant difference between analog and digital video is the fact that with digital video the video quality can be varied based on the bit rate used to encode the original video the video bit rate is given in units of bits per 2nd although it is usually in millions of bits per 2nd or megabits per 2nd the video coder sets the bit rate based on the target application and channel bandwidth that is available video coded under a fixed target bit rate restriction is called constant bit rate or CBR video in constant bit rate video the average bit rate over a short period is held constant this is accomplished by ensuring that each picture is either encoded with the same number of bits or if pictures are encoded with a different number of bits but still limited to the maximum number of bits the average bit rate over a short period as may cause an via stuffing any low bit rate periods with no bits to fill out the bit rate to the target constant bit rate value thus in CBR video each picture may be encoded with a different number of bits but the average bit rate over a short period is still held constant it does not exceed a target maximum bit rate pictures encoded with varying bit rates is called VBR or variable bit rate and in this case targeted average minimum and peak bit rates may be specified encoded bit rate also depends on the video resolution the dynamics of the video be encoded in the type and generation of encoder such as an MPEG-2 encoder MPEG-4 or the latest generation HEVC the high-efficiency video codec will discuss these encoding standards and a bit more detail later in this module now let's discuss compression of digital video raw digitization of video actually

increases the data rate or bandwidth required to transmit the signal but it turns out there is a little or a lot of redundant information in that raw uncompressed digital video since cable operators need a way of reducing the size of digital video they use digital video compression of digitized video to remove redundant information since different receivers need to know how to decompress the video they receive the MPEG picture experts group or MPEG which is an international standards body was developed to develop a standard for encoding and compressing video images called MPEG the name MPEG comes for the working group of experts that was formed by iso and IEC standards organizations MPEG compression formats take advantage of spatial and temporal redundancies in the video stream to achieve higher compression ratios spatial redundancy is within a video frame and temporal redundancies are between video frames jabber also takes advantage of motion there are 2 terms that are used to describe compression lossless and lossy the lossless compression herein lossless compression the decompressed digital video is exactly the same as the original as you can see on the left-hand side is very crisp and clear not discarding video data during compression the operator can reverse the process and get the same data the only problem with this method is that the file size is still rather large and then when we stream it to the customer it's still large to wash all lossless is not an option for cable operators but it was on the other hand you have all like a lot lossy or too lossy as of yet exact opposite of lossless's lossy word the regenerated video data is different from the original we can clearly see some lossy information where we lossy lost parts of the video because it was removed and subscribers definitely know this the differences and if they do the subscriber may notice what we call video artifacts so it is necessary for MPEG to use a lossy technique for SD HD and altar HD but not too much of it lossless would produce video bit rates it would be too large liver over qualm was I just mentioned us his lossy compression is used for standard digital video the qualities considered inferior to analog video compression factor will reduce the size of the digital video content without significantly affecting proceeds video quality but by the subscriber so as you can see that you can see different frames here and some frames or the entire frame and some his work content is removed and the compression factors achieve with MPEG is also referred to as the coding game MPEG is a lossy coding scheme many content is removed from the stream when too much content is removed subscribers will notice the artifacts that I mentioned in a digital video content there is an example in the screen of a art that everything about compression… Take a look at some of the sizes that we can achieve through compression that we know that if we had an HD would be about 1000 Mb per 2nd on but this is in a slide that actually shows the estimated bandwidth for network-based entertainment and given the service weather is an MPEG-2 video streaming it sees about 3 to 5 Mb per 2nd HD about 12 to 2015 be in the average ST at the HD MPEG-4 you have HD HEV C which is about 5 altar HD 4K which is coming whether we like it or not it's about 15 voice over IP is about a meg broadband date is about 5 to 10 and that's growing gaming is about 2 Mb per 2nd and audio streaming about wanted to given all this data coming into the customer's home MSO's feel that about 75 Mb of bandwidth within the home is necessary for service delivery and on the right-hand side we see the Cisco visual network indexed and we can see that each one of these areas how they are growing from 2009 to 2014 so you can see there's a huge growth of Internet video and that's that light blue shade right there and in the middle is a filesharing picking up just slightly might I continue to see cable VOD slightly growing will bit more business Internet growing a little bit more along with web data so this is only going to continue to grow compression reduces the amount of data required to produce a digital video image by reusing

pixels or other data this means fewer bits needed to be transmitted over the cable network less data equates to lower bandwidth or bits per 2nd MPEG-2 and MPEG-4 and advance compression schemes like MPEG high-efficiency video coding are available tools to compress video most cable operators use symmetric model of compression where the encoding and decoding of the same complexity in the asymmetric model MPEG a single compressor typically encoder fees a large number of the compressors is very common we have encoder and the head end and many many set-top boxes on the field another example of asymmetric is a DVD system compression utilizes the order in patterns found video information to reduce the size there are several types of compression techniques we talked about and sophisticated compression uses multiple techniques oppression is made up of several basic key areas call processing blocks we have the preprocessing the DCT or discrete cosine transform we have cues or quantization we might even have what we refer to is required is Asian is shown from the output buffer back to quantization we have runlength encoding and that combines that you might use related coding or variable length coding after quantization and assist in reducing those quantization quantization values even further and in the output buffer is the coded frame encoding is the process of preparing the video for digital let's take a look there's a big picture of encoding over the air satellite direct fiber or methods used to receive video by the cable operator encoders or transcoder's may be used after the receiver is a transcoder may be used to convert MPEG-4 AVC part 10 video streams in the MPEG-2 video streams carousel servers provide the necessary guy dated the set-top boxes for channel lineups the Azeris provide digital commercials to be inserted in the video stream multiplexing is used to create groups of channels multiprogram transport streams multiplexers assign multicast IP addressing with UDP port numbers for routing running is also used to pull video streams in the head and for modulation using edge qualms the network ring is used as a delivery transport typically for fiberoptics that connected the transport and that will connect all the areas of the cable network together for video delivery that ends in the hopes here is a simplified block diagram of a composite video encoder the preprocessing or video filtering of encoder is used to reduce noise in the input video signal remove artifacts like ringing from a previous filter and convert color spaces the encoder may include several filters that focus on removal video artifacts in addition to the reduction of video noise the preprocessor will lock encoding groups of pictures or cops to previous encoders if required and form pre-analysis use for the encoding process the video buffer stores compress data for processing the red green blue RGB color spaces may be converted at this stage to an input video color space of YUV or YC BCR luminance and chrominance artifacts are elements of the input video that do not belong in the video information the discrete cosine transform were DCT stage decomposes each 8 pixel by a pixel macro block into a series of waveforms with specific spatial frequency that's typical of an MPEG-2 the DCT stage out was an 8 pixel by 8 pixel block of horizontal and vertical frequency DCT coefficients shown here the quantization or Q stage eliminates the unimportant DC coefficients I was the lossy stage Rossi throwing out coefficients orally call the high frequency DCT coefficients one is Asian is concerned with limiting the permissible values needed to represent the DC coefficients this quantization adds noise to the video stream the encoder will perform a check on the quantified quantize coefficients in verse quantization or IQ stage computes the inverse quantization matrix by multiplying the quantized DCT coefficients with the quantized table at the Q stage inverse DCT or discrete cosine transport is the original input 8 pixel by a pixel macro block I DCT errors are expected due to quantization the frame memory buffer is used to store frames assessing typically 3 frames stored at this stage the motion

estimation or Emmy stage uses a scheme with fewer search locations and fewer pixels to generate motion vectors in in indicating the directions of the moving images the motion compensation or MC stage increases the compression ratio by removing the redundancies between frames called temporal the variable length coding or veal seek encoding works with a lossless entropy encoding process to reduce the bit rate required represent the DCT coefficients by sending shorter codes for common pairs water codes for less common pairs there are 3 types of entropy encoding uses used and their Hoffman limp L Ziv and arithmetic coding entropy encoding uses runlength encoding or zigzag scan method to remove repeated DC coefficients assigning each DC position with a pointer to its original value or to its value the most useful portion of the digital video signal is the entropy the regular or rate control throttles the bit rate by changing the quantization raws say France's 8 bit versus 10 bidding quantization for example using fewer bits per TC DCT coefficient might be one other option used or lowering the number of bits add more noise so you got a watch if you go from a 10 bit the 80 little bit more noise higher number bits adds less noise but is a larger file size the output buffer is the data buffer on the output of the encoder in a sexual take a look at the standardization for compression techniques using cable in particular MPEG or the moving pictures experts groups the illusion of motion is produced by rapidly displaying a series of still images the still images comprised of horizontal lines of video eliminated one pixel at a time when individual pixels or blocks of pixels can be reused either elsewhere in the image of the next frame they do not need to be retransmitted and this is the theory behind MPEG so instead MPEG can reuse these pixels on this will help MPEG reduce the bandwidth required to transmit the images and is also referred to as compression and this would MPEG-2 to do here in the 70s few engineers thought it would be possible to transmit HD due to video over a 6 MHz HSE channel compression of digital signals made it possible to shrink the size of digital signals like HD and other signals like audio operators leverage the moving pictures experts group or MPEG standard MPEG has many attributes of analog composite video MPEG is considered in some sense the modern digital equivalent MPEG offer superior audio and video quality at significant bandwidth savings as shown here digitizing content without the means of compression will produce an HD transport stream that are too large to carry over and HSE are outside plant MPEG encoding can reduce the bit rate of a video stream some of the compression factors we talked about phrases MPEG-2 50 to 1 MPEG 401 or MPEG HEV Sia 201 sampling rates are used to reduce audio at 41 and 8 to 1 compression factors MPEG audio like advanced audio coding or AAC supports up to 48 channels at a sampling rate of 96 kHz MPEG compression looks for redundancy in the video content via redundancy does not add anything to the video information being consumed by the subscriber MPEG compression is based on 2 main redundancy algorithms are spatial or intra-coating coating within each individual MPEG frame or temporal or inter-coding redundancy found across a series of MPEG frames called a group of pictures are God MPEG also allows an Emma sewer cable operator to transmit video content using MPEG-2 transport streams called framing MPEG transport streams carry video audio and data that is multiplexed together and are identified by program IDs called kids the MPEG transport stream is used to packet size video data into a 188 byte packets MPEG-2 transport streams are also used by MPEG-4 compress content here the set-top boxes a coding video and audio programs found it MPEG transport stream what it will do is I'll match the program IDs in each one of the programs and pull that into the channel that the customers watching MPEG technology has revolutionized the way cable operators deliver video services MPEG offers a cable operator several benefits superior digital video quality of substantial

bandwidth savings it's required to receive today's satellite digital video signals it's also utilized to receive signals at the content distribution network in an MPEG-4 format so MPEG is that there is well and is used in all types of services where as VOD or STB services MPEG is both a compression algorithm and a transport and MPEG-2 MPEG-2 supports a transport of streams independent of MPEG compression also cost savings for storage and retrieval of digital video program it's it's cheaper to store less content if it's compressed and then this for sending content if I could send it was bits is gonna save me money anyway further savings due to MPEG standardization everyone's using the same MPEG standard across the world that means it's better for interoperability among vendors of unbind MPEG product from vendor a it would be very similar or if not the same as MPEG B or others as vendor B compression is based on spatial or intra-coating without a frame and temporal or inter-coding cross frames redundancy waste we talked about that is very very important MPEG feature allows it to compress there are many digital video standards now from MPEG-2 to a TVC but there are also different digital video formats which can be either high definition HDTV or standard definition SDTV using either progressive or interlaced picture scans let's explore digital video standards formats and something called come pending next the advanced television systems committee or a TSC is a digital television standard that defines 18 standards the ATS C replace the analog NTSC television standard is cable operators move to all digital develop by the advanced television systems committee the high definition television standards was defined by the ATS C to produce the following widescreen 16 x 9 images or the aspect ratio there are 2 HDTV standard resolutions 19 20 x 10 80 and 12 80 x 7 20 a host of different image sizes are also supported more than 6 times the display resolution of NTSC a TSC boost theater quality audio using digital Dolby Digital AC 3 format to provide 5.1 channel surround sound and numerous auxiliary data casting services can also be provided the ATS C standard fits a 19.39 Mb per 2nd payload of a 6 MHz channel which is ideal for 8 VSP or a qualm channel that we using cable various SDTV formats are supported such as the international telecommunication Union ITU RDT 601 – 5 format the videos compressed using MPEG-2 Dolby audio codec 3 or AC 3 was chosen as the audio compression scheme and uses the Society of motion picture and television engineers standard 5.1 to a code 5 audio channels and limited bandwidth subwoofer ancillary data is also included in the standard Dolby AC 3 transmits 5 audio channels for left front right front front center left surround and write surround along with a narrow beam with channel AC 3 is ideal for mono stereo or full surround sound the advanced television systems committee ATS C standard is defined in documents a 53 part 1 to part 6 describing the system characteristics of advanced television or digital television systems a guide to the use of the ATS C digital standard is provided in a 50 for a and is shown here in the upper right-hand corner of my slide while digital audio compression light Dolby AC 3 standard is covered in a 52 these documents may be obtained in the SCT portal under course materials some of the names that you may here for advanced television formats include digital TV or DTV standard and this is a standard definition digital TV or SDTV is high definition TV or HDTV ultra high definition TV or UH DTV or even enhanced definition TV or eating TV NTSC analog television is not advanced TV format since it's original analog TV format was used in the United States and North America and neither his pal the European analog TV format the let's talk about how the ATS C standard is used in cable aspect ratios relationship of with the height of a television screen a TSC uses different image aspect ratios than the more common national television systems committee NTSC measures for units by 3 units work we refer to as 4 x 3 the NTSC dimensions originally were chosen to match 35mm

motion picture film the 1.33 x 1 aspect ratio wider a TSC aspect material HDTV is 1.78 by one or 2.35 x 1 films these must be reformatted for viewing on 4 x 3 sets the types of reformatting use currently include payment scan where a human manually moves a 4 x 3 window around a 16 x 9 video to follow the most important action of the screen note that in this case some of the screen information or view is inherently lost in the process we have letterboxed or black bar for the entire 16 x 9 images shrunk to fit in a 4 x 3 screen and black bars are placed around below the resulting image to fill in the gaps left behind and we also have anamorphic squeezing where the video images squeezed in a nonuniform way in order to store or transmitted over a medium with ace different aspect ratio and then re-expand it before displaying it this process was designed to make maximal use of the pixels in the storage medium which originally were 35mm film even when filming widescreen format video but it also use the storing or also used for storing widescreen formats on DVDs for example we use this format: anamorphic squeezing there is a slight distortion of objects from the original video image using this process but it retains much more of the original video resolution and quality than the payment scan or letterbox approaches the term anamorphic is from Greek and it means forms again there are several standards that are available MPEG 1234.10 VC 1 and a GVC of MPEG-1 standard is not really used in in your notes you can get some additional details of the ITU standard of the international total medication standard in the iso the international standards organization and some details are but but MPEG-1 was a VHS and then and VCD standard MPEG-1 for a lot of pixels away so it really wasn't optimal for cable operators MPEG-2 yes that's that's very very common in in cable support oppression based ESD 3-D juice on DVDs as well and it also has a audio standard will look at little bit big 3 there is no standard was found to be redundant all the features of MPEG-2 so was never standardized so don't worry about MPEG 3 MPEG-4 part 10 is what we refer to for cable operators is sometimes called for about 10 it's also a lie to you and iso standard is known as advanced video coding support oppression of HD 1080 P formats and IP TV and SCV there's also another one called VC 1 that's a 74 to 420 1M standard it's used to compress a blue rain sing a Slingbox video content of a not found too much in a digital video transport or digital video content within the cable operator we we stick to MPEG-4 part 10 and we also have MPEG HI which is also known as H265 which is a high-efficiency video coding and this is designed to support oppression of ultra HD formats like for K and H a K television which is coming so it just a little bit know that standards the iso is the international standards organization I mentioned the ITU is the international total medication union the IEC is the international electro technical commission of these are all the groups that work along with MPEG to create the standard there are additional extensions where added to MPEG by the digital video broadcaster DVB standards and also the advanced television systems committee organization the ATS C is the standard way we receive video over the year and so they also have standards on everything from aspect ratio and and in the formats we use 1080 P7 20 P in the resolutions that so here's MPEG-1 there's a lot of details in the notes but you see a lot of the standards that were created for MPEG-1 it was started back in July 1989 as H261 it was designed to produce reasonable quality images but it has a very lossy at 75% of the video pixels is that is dumped it's very common for a VHS and stereo at 192 kb per 2nd MPEG-1 is lossy we talked about that uses facial or intra-coating of frames is to take advantage of temporal style is a standard 1994 MPEG-2 which is also known as H262 is is the center for compression for ASD HD and VOD and many other forms of video the cable operators use we can work out without the bit rates up to 40 Mb per 2nd so that's great most HD is not going to go pass 20 Mb per 2nd so

that I work just fine with sever for K and a.k.a. nets were working to get into MPEG H and maybe some MPEG-4 as well MPEG-2 takes advantage of the spatial and temporal passes between the tool makers of the 50 to 1 ratio again there's additional things in the notes here that you can read through about MPEG 2 MPEG freely I said it never materialized as a standard so you can fill in the notes of some of the details but MPEG 3 was designed H263 was designed for for HDTV but it was dropped because MPEG-2 was was able to do that finalize in 1998 MPEG-4 ITU – TH264 compression format allows deeper compression of digital video using objects using audio to video objects allows MPEG-4 to improve lossy compression over MPEG-2 video codec VC 1 which is also known as 7420 1M and other advance Kodak's are new formats which include even more sophisticated tools and MPEG-2 to achieve higher compression ratios such as 100 to 1 without losing image quality MPEG-4 is now being the court becoming the industrywide commercial appointment version of MPEG in the cable industry the main focus of MPEG-4 was a range of low bit rates from 384 kb to 8 Mb under MPEG-4 part 10 which is what's deployed in cable this is functionally added for cable environment supporting 38.4 Mb per 2nd profile level and a studio 1.2 Gb per 2nd profile level MPEG-IV is based on the MPEG-1 into standards and virtual reality more of modeling language or VRML MPEG-4 as the following to MPEG-1 into an VRML support for 2-D and 3-D content support for several types of interactivity coating a very low bit rates are 84 kb per 2nd to very high ones 1.2 Gb per 2nd it has standardized DRM or digital rights management otherwise known in the MPEG community as intellectual property management and protection are I PMP has native support for natural content in real-time stream content using URLs several forms of support over IP and other networks with bandwidth unknown at the time of encoding supports the advanced audio codec or AAC this was standardized as it is a adjunct to MPEG-2 as part 7 before MPEG-4 was issued while these advanced features lead to better compression ratios while maintaining video quality MPEG-4 does not require more processing power and a decoder so for example the set-top box or PC or tablet client application doesn't need to have additional processing note that I mentioned MPEG 7 and this is MPEG 7 was intended to provide complementary functionality to MPEG standards above MPEG age or ITU – TH265 high-efficiency video coding standard is most recent joint venture video project with the ITU – T video coding experts group VC EEG in the iso IEC MPEG standardization organizations working together in a partnership known as the joint collaborative team on video coding or JC TVC the 1st edition of the AG VC stander was to be finalized in January 2013 resulting in a aligned tax that would be published by both the ITU – T in the iso IEC additional work is planned to extend the standard to support several additional application scenarios including extended range uses with enhanced precision and color format support scalable video coding and 3-D stereo multiview video coding in iso IEC the AG VC standard will become part of the MPEG – H part 2 iso IEC 23 008 – 2 standard and it is the ITU – T it is likely to become an ITU – T recommendation of age.265 the resolution of MPEG H is 3840 pixels by 2160 pixels for the 8K is 8,294,400 pixels or 8.2 megapixels while these advances lead to better compression ratios while maintaining video quality MPEG H will require more processing power in the decoder than MPEG-4 or MPEG-2 here we CST full HD and ultra HD formats compared BST starting on the bottom left offers 720 buys 576 or 640x480 formats because he is a very small resolution full HD offers 10 80 x 19 20 progressive so much better picture and this is what our customers have become accustomed to now we have 4K standard which is another version of HD at 21 60 x 40 96 progressive so much larger picture much better resolution them overseeing with HD and we now are even hearing about some 8K ultra HD offering above 43 20 pixels or

4300 pixels by 7680 pixels progressive took as a good comparison here of the different formats and and where video could go was transition and talk a little bit about audios of this section works for some of the moving pictures expert group and digital audio you think you sent recap video compression MPEG compression removes redundant information after digitization of the original video and then can further reduce the bit rate by filtering off less noticeable information such as higher spatial freak frequency content from the video image which is to say reducing the level of detail in the video image compression is used to shrink the size of digital video formats and resolution to fit into a 6 MHz radiofrequency channel for over the air 8 VSB or access network Quan transmission the MPEG transport stream is used to pack a ties video data into the 188 byte packets and MPEG transport stream will have a header that is used along with user datagram protocol Internet protocol and data over cable service interface specification to transport the video over the cable operators network but what about audio how is the audio bit rate compressed or reduced similarly to the digital video compression audio can be compressed by removing redundant information but also by filtering off higher frequencies or less noticeable frequency harmonics but even before digital audio compression we use something called compounding which stands for compression and expanding as a way to process analog signal on a logarithmic scale in a manner that improves the signal-to-noise ratio in a different signal of an analog stereo signal and we use this concept in digital audio sampling and compression as well which will explore next audio could be synchronized with video or could be standalone the MPEG packet ties elementary stream were PS in a single program transport stream contains both audio and video MPEG-1 audio is a 2 channel with sampling rates of 44.1 kHz 48 kHz 32 kHz and compressed bit rates of 32 to 192 kb per 2nd per channel MPEG-2 backward-compatible is a multichannel of the 5.1 with additional lower 1622.05 and 24 kHz sampling rates MPEG-2 AAC or advanced audio coding is a very high quality audio coding standard for 1 to 48 channels at sampling rates of 8 to 96 kHz with multichannel multilingual and multiprogram capabilities let's see how this works human hearing is limited to frequencies lower than 20 kHz in most cases above this range most human ears are insensitive although other animals like bats can hear above 100 kHz in fact that is the key to the accuracy of their sonar but since human hearing is limited to 20 kHz that limits the amount of information we need to sample to accurately reproduce the audio but the hearing of older humans is generally less sensitive to upper frequencies in fact the degradation starts after the teenage years so one way to reduce information content of the sampled audios to filter the audio even less and 20 kHz say to 18 kHz most older adults probably couldn't even tell the difference although teenagers might be able to but says it doesn't apply to all humans is not a generally good way to reduce the audio information content however human hearing also turns out to be insensitive to softer frequency components in sound that accompany louder frequency components in the same sound in this works for all humans regardless of age removing the softer frequency components in the presence of louder frequency components doesn't significantly significantly change the perception of the sound by the human and removal of information means we can use fewer bids to transmit that information this is the basis of the MPEG free and other digital audio compression algorithms to remove frequency component is a don't really change the way the audio sounds humans this process is depicted on the slide if you're sampling the original audio lower amplitude frequency components are removed therefore therefore reducing the information content in the audio and sauce reducing the number of bits required to store and or transmit the audio information another way to reduce the bit rate of audio is to sample the

audio on a logarithmic instead of a linear scale logarithmic scale sampling offers a similar level of accuracy with fewer bits than when you're sampling combining all these techniques together we find it commonly found compression ratios for audio or for the one or up to 81 sounds like a lot compared to video which can be compressed as much is 50 to 1 due to the great amount of redundancy or redundant information and video compared to audio plus humans are less tolerance are less tolerant of audio over compression since it has much less information begin with essentially the narrow bandwidth of audio makes higher compression ratios perceptually in top intolerable Dolby AC 3 is not part of the MPEG standard although it can be carried in MPEG-2 transport stream is a private datastream Dolby AC 3 has been adopted by the IATSE or for HDTV North American digital cable uses the IATSE format as well and that allows us to use the Dolby AC 3 format most set-top boxes now support MPEG-2 audio and Dolby AC 3 audio and is part of the DVD audio spec along with MPEG-1 and to audio in linear PCM the compression algorithm is proprietary in nature and must be licensed from Dolby labs Inc. the compressor bit rates offered by Dolby AC 3 is 32 to 640 kb Dolby AC 3 supports Mono to full 5.1 channels the 5 channels are mentioned earlier left center-right left surround rights around the .1 channels is a subwoofer little factoid here in the notes most fees found in the cable had enter off satellite are encoded AC 3 with a 48 kHz sampling rate the bit rate of 192 or 384 kHz the ATS C is a is again as the standard for over the air a 52 standard defines 2 ways to create coded representations of audio information how describe these represent representations how to arrange these code representations for storage or transmissions and how to decode the data to create audio the code representations defined are intended for use in a digital audio transmission and storage applications ATS C8 52 is how AC 3 and enhanced 83 are applied to the digital video broadcast digital television standard audio engineering Society 3 or AES 3 is a standard used for the transport of digital audio signals between professional audio devices finally note that he had enter hub tech you probably will be changing the audio compression ratio in encoded content those decisions are typically made now the corporate level and are based on extensive testing with encoders and video audio quality assessments but you will be using wiring to connect audio sources and receivers together so just keep that in mind that just like the video over collection had ends which uses either 75 or 50 ohm impedance cables who transported Ned and audio cables in the head and and associated equipment or standardize and typically have 600 ohms balanced impedance and use of a nonstandard audio cable can lead to problems just like in nonstandard RF collects cable in this module we discussed

7.

What is the overview and STB components module this module is intended to provide participants with a general understanding of switches will video fundamentals upon completion of this module the participant will be able to state the difference between broadcasts and switch broadcast explain how different services are placed on the cable transport identify examples of bandwidth constraints faced by MSO's describe how STB provides a solution to MSO bandwidth constraints also in this model to be able to identify the system sizing and bandwidth requirements described channel candidates for switch digital broadcast illustrate switch broadcast economics and time for plant upgrades describe switch broadcast operation as you probably already know MSO's are offering numerous services cable customers in general they can be placed into 2 categories broadcast and switch services broadcast services include both video and audio services for analog and the digital transports are always available for viewing

the 2nd category includes switch services that unlike broadcast services require a two-way downstream and upstream path the subscriber can then request the service or activation of some type these services include VOD with the viewer can request a program service thereby switching in a dedicated adjuster into the set-top box will also start over which the viewer can request to restart services after program has started and is still in play and switch broadcast or STB which is the theme of this learning session which also requires a return path where the viewer request that switches in one of the inactive broadcast services for active viewing Seymour to show what you currently watching here's how you do it is the beginning of the program just press the select button at any time during the program regardless of when you can do as long as you see start over as an option to go back to the beginning of the show catchy theme song for free you the spontaneous time that you go back one to 3 days to catch the shows you missed without even setting your DVR so you can your schedule off the TV schedule just press the select button on your remote 12 for more of the channel you watch scroll to the back options in the menu to view a list of available programs and select the show you want anything from also switch to the HD channels available at the touch of better now and make sure to check out her other helpful videos to help you get the most out of your TV Internet and phone experience is review the cable industry at one time only carried analog video sources analog televisions then were needed and are designed to tune into services between the 52 550 MHz range for cable TV when the viewer selects a program channel the TV turns to a center frequency in the analog spectrum that uses a 6 MHz wide path for transport since each analog program services title 6 MHz wide channel on the spectrum there is only room for 82 channels this limits the number of program services to 82 program sister is a one-to-one ratio between program services and channels that is to say each analog standard definition program service requires one 6 MHz wide channel in addition to the limitation of the one-to-one analog service to channel ratio that limits the overall number of available programs analog services are not suited for deep transport since signal degradation occurs over each hop of the transport however since analog televisions are simple to use and designed to tune in directly to the fixed center frequency for each program set-top boxes are not required in the mid-90s digital TV became a reality to consumers and what services were digitized and later compressed and uncompressed video stream approximately 270 Mb could then be compressed into approximately 3 to 4 Mb for standard definition video stream in general 10 or more compressed SD streams can fit into a Quan 256 modulator providing digital throughput of approximately 38 Mb per 2nd to 86 MHz wide channel on the transport there are 10 or more digital streams on each 6 MHz channel when a viewer selects a program channel set-top boxes required to associate that selection to a frequency on the spectrum so you can pick out the relevant digital stream to the MPEG decoder before outputting it to the TV HD services require additional bandwidth compressed HD services offering the 10 ADI resolution with a 16 x 9's ring ratio are generally compressed somewheres between 8 to 20 Mb per 2nd for example each stream is clamped at 15 Mb then to HD streams combine for 30 Mb per 2nd will easily fit through a Quan 256 modulator capable of 38 magnets per 2nd switch broadcast supports a mix of both HD and SD streams on a transport so generally to HD streams in a couple of SD streams will easily fit on a Quan 256 frequency if HD services are clamped at 12 Mb per 2nd and one can expect to fit 3 HD services within the 38 Mb per 2nd Quan bandwidth however compression down to 12 Mb will often affect HD quality and defeat the purpose of HD content generally with today's compression ratios of 8 to 20 Mb per 2nd is, to clamp the HD streams to 15 magnets per 2nd this

allows place in a combination of 2 HD and a couple of SD streams per transport data services are also placing the man's in the cable spectrum generally North American MSO's are using a 6 MHz channel 1 256 for their downstream path for approximately 252 400 subscribers with showing assuming that not all subs are connected at the same time using the 250 to 400 subscriber model on the upstream path is, to use a Quan 16 device 48 Mb of traffic but it demands another 3.2 Mb from the progressive spectrum with a release of channel bonding than your cable modems will be able to connect to multiple Quan to aggravate the download down with bison even more demand on the frequency spectrum was North American MSO's have already expanded their analog 550 transports to support digital services after the 750 MHz range digital simulcast generally eats up 7 channel or frequencies in the digital domain assuming you can place about 12 digital programs on the 7 digital channel frequencies you can replicate all the analog services on that digital spectrum the remaining digital frequencies are typically used for additional broadcast services and narrowcast services such as peg public education and government channels or video-on-demand services there is also a demand for HD television on the doorsteps of most MSO's so there's a real need to find a solution to expand space on the transport systems now some MSO's have already opted to expand from the 552 750 MHz plants have decided to go even wider with 860 MHz or even 1 GHz size plants but this option is very costly in both time and money upgrades to the water plants often require rewiring the cable infrastructure and new equipment is often needed to handle the water plants the need to expand services on the digital transport begs the question whether a better upgrade solution exists there is an alternate solution and that solution is which switch broadcast will allow you to expand services in an already congested network in a relatively short time by switching and only those services used by the service groups switch in Israel and available now for deployment turnaround time for deployment is rapid is big and has often shown that it deployment from head into edge can be done within 90 days from order as was mentioned briefly in the previous section broadcast service do not require any return paths from the television or set-top boxes video and audio flows unit directly to the set-top box all program services are dedicated to fix frequencies a program selection causes a tuner to turn directly to the frequency assigned to the program VOD start over and switch broadcast services all require a return path to the head and since the services are all requesting unique nonbroadcast type of services there are numerous types of set-top boxes in the field and most bidirectional set-top boxes use a key PSK return path however boxes return paths via CMTS systems have also been used successfully for switch services the advent of two-way cable card also shows promise in standardizing a return path for all set-top boxes the slideshows the start over process each broadcast program is spool to a hard drive during the broadcast lowing set-top boxes to restart services anytime during the broadcast the start over service requires a return path from the set-top box to restart the broadcast session as a unique POD session to the set-top box the slideshows the relationship between the broadcast and start over service the top yellow bar shows the timeline of the broadcast service from 8 PM to 9 PM the pink line represents the request from his set-top box at a 10 that establishes a unique unicast flow from the VOD server to the set-top box once the broadcast is complete 9 PM roughness by the yellow line the service will no longer be spool to storage and start over process cannot be used on that program switch broadcast services also require a return path is illustrated in this diagram generally low-pass type lecture is used to pass through the key PSK frequency to a key PSK demodulator the demodulator then passes on the request over ethernet and IP to the switch broadcast session server that is often referred to as the SPSS

the switch broadcast session server or SPSS indicates to the switch in qualm to ensure that the requested program service is assigned to a Quan frequency notice that not all services are switched some programs are dedicated to a Quan is illustrated in this diagram by the Quan labeled number 1 the switch in qualm while number 2 is dependent on the SPSS to switch services on and off the Quan program services assigned to Quan number 2 are designed to exceed the qualm's capacity with the assumption that not all program services are being watched by users of a given service group although there are many different ways to layout the qualm spectrum this telegram illustrates how MSO can mix the array of services on Quan's using switch broadcast on a 750 MHz plant to increase number of HD services in this example 82 program services are signed to 82 Quan frequencies in the analog domain the same 82 services are digitized and replicated on another 7 Quan frequencies for digital simulcast then we hundred 20 ST services are allocated to 10 Quan frequencies for expanded digital services then the remaining 16 Quan frequencies are reserved for switch broadcast services was 16 Quan frequency assignments the MSO can place on the switch broadcast platform 32 active HD and 32 active ST switch services assuming 2 HD and 2 ST services on each qualm however at the service group is size down and with switch broadcast capability to oversubscribed Quan's you can potentially have unlimited selection of switch program services as viewers will only be watching a portion of the overall switch channel guide at any one time in summary switch broadcast allows you to free up a good 50 to 80% of your digital spectrum for the potential of unlimited services it fits seamlessly into existing cable architectures with minimal impact on operations a relatively short time typically 3 months. For the 1st tub to be up and running key steps for deployment include preparing her head and content for switch services defining your switch broadcast service group size and downloading switch broadcast client software to the set-top boxes switch broadcast has already been successfully deployed over 7 million households and ready for you to use now this diagram shows you the differences between standard broadcast and switch broadcast bandwidth usage at the edge the standard digital video system will have all the program services from the head and transported to the edge every services dedicated to a Quan and available for the set-top box to turn to switch digital systems still transport all services to the edge with one exception no services are placed on the edge forms until requested by the set-top box services are only activated on the qualm's as a set-top box to store program service unlike VOD the services are broadcast services so for 2nd set-top boxes the same program this year the screen on the qualm and no extra bandwidth is needed since there are typically 500,000 tuners in the service group only a fraction of the available services are activated therefore saving considerable bandwidth over time on what services can be swapped out with newly requested services keeping the qualm usage to a minimum despite the large number of services available to the subscriber this chart provides a summary of concentration ratios collected earlier to help compare and determine the best service group size generally to take advantage of switch broadcast services you should target a concentration ratio of 3 to 1 or higher concentration ratios can be adjusted by changing the number or type of service and a service group or adjusting the number of tuners on the service group one other observation or supposition we did notice was that some of the program channels were always being watched and never switched then made sense to move those popular program services that were always being watched out of the switched service and into the standard broadcast domain and other factors demographics the some areas may differ in program popularity these trials occurred in Texas views were inclined to watch the western channel such as the Oncor Western the family

and action was services yet those same Western channels may not have the same high popularity rating in areas such as New York or Florida generally if a channel receives a popularity rating of 60% or higher that should be considered as a candidate for standard broadcast program services with a popularity rating below 60% should be considered as candidates for switch broadcast in summary the switch broadcast trial succeeded big-band networks expectations the trial proved that switch broadcast could offer MSO's an alternative when deploying additional services on their networks since the cost per household is significantly lower than the plant upgrades at significantly improves qualm density and the number of services available to the service groups are increased dramatically finally switch broadcast is a proven technology and implementation is now a turnkey operation required about 3 months to run switch services from head into the 1st at Chubb this very high level block diagram illustrates the head and in some of the etched components be hidden components typically includes a digital multiplexer to groom and clamp incoming services to a single output that switch broadcast environment ST services are typically clamp best regards 75 Mb per 2nd and HD services clamp at 15 Mb per 2nd all switch services are then IP multicast out on a cookie transport to multiple hubs each hub will have one or more switch and service to manage and switch services on or off the way of switching qualms the switch and server will also manage and track active and inactive services on multiple service groups in this module we discussed the difference between broadcast and switch

8.

Welcome to the digital troubleshooting and repair process model upon completion of this module the participant will be able to describe the different types of tests used to troubleshoot and repair digital cable services understand were digital errors occur in the cable network and review the troubleshooting process but the digital test section what are the digital tests we have at average power measurement, uses a metric for digital signal levels in measurements and cable television average power measurement is the RF power expressed in terms of voltage defined as decibels relative the one millivolt where 1 mV equals 13.3390 W in a 75 ohm impedance mathematically the MDB MV equals 20 log 10 VX divided by 1 mV where VX is the voltage in millivolts that is the be converted to DB MV another test we have is tilt tilt is the amount by which the cable signals frequency components are not passed through the cable network at equal levels higher frequencies are attenuated more by the coaxial cable in the lower frequencies this effect is countered by providing greater implication to the channels at higher frequencies is important to maintain an operator still another test is qualm analysis and constellations test equipment that can graphically display and analyze a constellation patterns of a digital qualm or PSK signal more commonly known as digital signal analyzer or qualm analyzer and modulation air ratio or MER in digital transmission the ratio of the average signal constellation power to the average consolation error power expressed in decibels cable modem set-top boxes and sometimes equipment call this parameter signal noise will see that the noise floor in channel frequency response including able to tilt and ripple group delay variation and micro reflections oscilloscope or oscillator phase noise receiver imperfections and all other impairments that affect the receive symbol consolation can degree murder and continuing on with our digital test we have the bit error ratio or BER the usually pre-and post test that uses FEC forward error correction and also the BER 202nd test BER is the ratio of the number of arid bids received to the total number of bits sent in a digital communication system maintaining burrs

extremely important to digital communications bid errors are actually corrected by the receiver processing known as host BER using a process known as forward error correction on a test of spectrum analysis since both the downstream and upstream cable signals are sent in frequency division multiplexing or FTM the RF spectrum analyzer is arguably one of the most critical pieces of test equipment in cable facilities and in the field for troubleshooting RF issues in the network reverse ingress another test and ingress test is the return in the return uses the spectrum analyzer function of a signal level meter to scan for upstream or return Path interference that can cause excessive bid errors in the upstream signaling from the customer premises we also have the docs is smack layer via the reference cable modem the cable modem statistics test provides comprehensive information for all the upstream and downstream docs of signals in the bonded set some of the final test we have the set-top box diagnostics and cable modem diagnostics or EMT 8 a diagnostics many of the CP devices issued by the cable operator will support onboard diagnostic screens these diagnostic screens should be used as a guide to troubleshoot as a guide during the troubleshooting process and should never be used as the only tool to troubleshoot RF signal problems codeword error rate a method used of measuring or cackling and downstream upstream codeword error rate based upon the total words correctable codewords and uncorrectable codewords signal leakage in cable talk indications this generally is the undesired omission of RF signals are signal leakage from the agency not all signal level meters are alike since there may be maintenance and service level versions available a maintenance meter may offer a full qualm consolation analysis while I service meter will offer a light version of the same tool where do digital errors occur while a lot of them happen at the subscriber so let's take a look at some of these has the customer created the errors one question we should ask why read the subscriber user CPE installed correctly has a of the gun the proper configuration the early using the proper cabling made from the set-top box of the television is it all video displays or a single video display here for high speed is it all the vices in the home or a few of the devices are just a single device I what channels affected for digital is it happening in particular time of day our neighbors affected is the interior wiring using RDs 59 does he install it like how does a drop from the tap to the ground block working pedestal the robot look other signs of damage water corrosion within the drop system will drop pass the reverse ingress measurement are all splitters rated for 1 GHz or 1000 MHz with a fitting snugly the proper fittings is a ground block installed correctly and bonded how does the wiring look in the knit no copper exposed and binding post disconnected some the questions to ask because this is where a lot of errors digital errors can occur without digital errors in the outside plant transmission medium actors and passives you have been coacts bid templates loose fittings damaged or missing end of line terminators damaged or missing chassis terminators on directional coupler splitters or multi-output amplifier unused ports loose center conductor seizure screws unused have ports not terminated thus especially critical of low value taps on use drop has supports not terminated use of so-called self terminating taps these for DB2 boards a DB for ports and 11 DVA port and is at the feeder and of lines such taps that act like splitters and do not terminate the line unless all airports are properly terminated Cantor damage cable including crack cable which causes a reflection and ingress defective or damaged actives and passives water damage maybe ice has gotten into the device water-filled cold solder joints corrosion loose circuitboard screws etc. some traction filters have been found to have poor return loss and in the upstream especially those used for data only service errors may also occur to head an in hub site that is currently beyond the scope of this material what are acceptable

metrics for to be 6 qualm digital here's a list of metrics that may be used consult your operator for specific metrics but we have upstream signaling noise in all its upstream is not I used to but is is qualm upstream will use the 16 or 64 qualm but what is ethical the noise look like is a 31 DB a greater downstream signal noise into precise Guam 35 to be a greater upstream transmit greater than 30 DB MV and less than 50 D BMB downstream receive greater than negative 8D BMB and less than positive a D BMB Amish leave 0D BMB is the FCC optimal upstream codeword error rate less than .000003 or downstream codeword error rate less than .0000003 failed cable modem termination system range request call T3 request that you be lesson for field keep alive acknowledgments also noticed T4 this lesson one downstream modulation or ratio 35 DB are greater at the subscriber 36 DB are greater the tap 37 DB are greater at the line extender amplifier Bridger 38 DB are greater at the node and 40 DB are greater at the head in her hub bit error ratio 1 EE to the -9th or less as one error in 1 billion bits care to noise ratio 30 to 35 DB mean opinion score greater than 3.6 and in our Savior if you're troubleshooting digital voice services things like latency become a factor was that less than 150 ms jitter lesson 40 ms packet loss less than .1% and signal leakage is 20 µmol per meter at a distance of 3 m obviously if we have leakage coming out egress of the plan that we have a potential for ingress and noise will affect our digital signals important aspect understand that leakage is the source signal as we move from analog channels to digital channels is important to understand how this will impact her ability to detect the associated leakage the difference is that digital signals are generally carried at a lower power level compared to analog digitally run at 526 DB MV lower in amplitude than analogs in this example shown a typical reading from the storm you will verify this this is an important point as the resulting digital leaks will also belong to the lower power as compared to analog leaks this must be factored in when measuring digital leakage and correlating it back to an equivalent analog week the SEC says analog limits on leakage at 20 µmol per meter at a distance of 3 m here we see a good reading and a poor reading of signal leakage this is an example of how a single leak location can actually be 2 separate leaks signal leakage degrades RF signal quality is important because customers become frustrated when experiencing poor picture quality week one is a high-frequency leak that was nondetectable to low-frequency and leak 2 was a loose connector that the boxes can be used to discover digital health here are the Motorola diagnostics the approach of the Motorola has towards diagnostic troubleshooting is a simple and effective way of verifying where problems might exist access to the screen is easy it can be done with a remote or with the 2 buttons on the front of the box for a Motorola set-top box use the sequences to enter the diagnostic screens turned the box off using the power button then press select within 2 seconds the diagnostic screens will appear information to use for digital troubleshooting is EP 00 will show connected doesn't replace the box the re-modulation channel is outputted on channel 4 is the common channel that if organ of this up using color exits were it would be to provide out of band status out of infrequency signal will noise a low value that we want to see on the automatic gain control will give us in band status will provide the mode such as qualm in the state in this mode a 5 2nd error count showing uncorrectable's and collectibles will also show the type of channel analog or digital the encryption orbits not encrypted weather was authorized the box are not authorized it will also provide or of modem statistics power level interactive status IP addressing and the state which should be in receiving mode or running mode for a Scientific-Atlanta setup box use the sequences to enter the diagnostic screens you press and hold the center cantle the messes LED lighting diode on the front panel blinks and then release taxes diagnostic screens present

diamond key for the info key now in the notes you'll find out the correlation between the model numbers in the diamond key and the model numbers in the info key in here and navigate the diagnostic screens repressing either the vole of volume up or volume down key is a diagnostic table is like you to view the actual channel with the diagnostic screens is a blended image this allows for excellent real-time troubleshooting capabilities to display blended image for troubleshooting purposes press the center key for the 1st blended level and will give you a dark blended level can continue to press the center key to scroll through the 3 blending modes dark like non-to exit diagnostic screens press the diamond key or the executing that here you can monitor power levels that white good amber okay red bed data geek and you can also monitor that what is connected and that's the beauty or is ready for broadcast only a means not doing the beauty word if there's an error like a slow boot weight error you also have the 4 data channel which is the other band channel the returned data channel which is return Path tuning mode to be 6 qualm is what we would expect IP address information the status at the locked there's also an unlocked mode and seconds how long the tutor has been running there's also corrected bites and uncorrected blocks and signal noise measurements you can gain from here if the set-top box can do this so can cable modems and MTAs and EDTA's oculus have their own diagnostic screens that are available on most manufacture cable modems toxicity screens on any given cable modem you enter when any 2 168 100.1 in the Internet browser of the computer that your modem is connected to most notable modem manufacturers have this page accessible and you can look at and those on which manufacturers have the what you can access the page it will give you some pertinent information on the status of your digital network information that is most important is a downstream signal level power which should be around for cable modem they 72 positive 5 upstream transmit power positive 35 to 50 DB and downstream signal noise about 30 DB are greater and consult your operator for exact levels and of course we learned a lot about this are ready and just to reiterate this this let you know that is important to follow good craft and ship all the time versus maintaining a center conductor length of 1/16 or 1/8 of an inch past the end of the F connector or following the bending radius of collapse which is 10 times the overall diameter by using the correct fasteners of your fastening collects somewhere within the customer's premises and install proper drip loops and fittings you here we see a bad fitting on the screen we don't use hex connectors anymore and a gazebo to get some bad bending radius going on there as well but all these things are very important to maintain digital quality and consult your broadband training modules for more information here in this session we will review the troubleshooting process tried to find the process used for troubleshooting service failure as shown here analyze isolate divide and conquer revolve and confirm for our example will discuss the resolution of video service failure however the process outlined below or the on the coming slide to the process will be used to resolve all cable service failures during troubleshooting keep the customer informed of what is being done and why you're doing it please be sure to follow all safety precautions during the troubleshooting process physical safety precautions when going into attics and crawl spaces and the like an electrical precautions like using your foreign voltage detector before handling faulty equipment remember safety 1st look at step 1 analysis of the symptoms as previously mentioned the 1st step in troubleshooting is to gather intelligence from work order and the customer to determine the areas of concern you'll be able to observe the condition of the drop in outlets and cable prepare problems to acquire the customer's perspective is important to ask open-ended questions about the field services keep in mind that most cable customers are not experts and may be unable to describe the

symptoms accurately using the cable jargon or acronyms if this is a video problem example at the customer of the issues present at all times and is affecting all channels using this information collected is easier to determine the failures a subset of chattels in the video problem example let's consider the following it may only be HD high definition channels or analog or just VOD content is having problems try to isolate which areas have those issues analog raise the general problems in the absence of standard definition digital video problems may be that the arsenic level is slightly below specifications when VOD content is the only issue is often point to an Austrian return Path issue is a failure only on channels that are part of a single multiplex that means are they part of a single qualm remember that we multiplexed many channeled into a single qualm or 6 MHz channel filters multiple channels attics could lead to one particular frequency have an issue with certain times of the day that the problem presents itself does the problem occur more frequently on hot cold or rainy days remember that the temperature does affect the attenuation of the collects cable and drop system has a customer had this problem the past is the present or more than one outlet TV or other knowing of the prom occurs on more than one outlet is important but remember the same problem will present itself differently on analog versus digital channel such as e.g. arrestee and also in outlets with or without digital converters or set-top boxes for example low signal level for the tab will manifest the snow on analog channels it may manifest in microblogging were no picture at all remember microblogging combined call tiling your pixelation on digital channels this is this is due from packet errors or in extreme cases of this have a frozen screen again from packet loss in the customer to original equipment with issues have the customer demonstrate the video problems that they can try to locate the failure when the customer is unable to replicate the concern you may recognize the problem immediately based on information presented later in this module you will quickly solve it if not recall recall back to the OSI layer discussion above that a key part of the 1st step is to determine what the physical layer looks like in this case the agency plan premise wiring make sure that these 2 pieces are operating correctly start with verifying the RF signal at the input of the DTA digital terminal adapter or set-top box may be that there is no box and you just verify at the wall plate we will make sure that they fall into because company specifications check the frequency specified by your cable operator refer to the appendix with tables for examples of frequencies to measure the downstream RF levels are correct the problem may be simple to DTA or set-top boxes unplugged or has a power issue has dead batteries and remote control even in the case of the latter a good practice is to check test and confirm that all services are properly operating in the customer satisfied prior to leaving the premises if levels are not within specifications or the services include not function properly the next step is to isolate the problem who step to isolating the source of the problem in the downstream RF signal level is not within company specifications and you have at least one problem with the physical layer and somewhere on the colossal network between CPE and that the problem must be addressed before proceeding any further with the use or with the troubleshooting of of customer premise equipment used to divide and conquer method of the next section to determine the location of the radiofrequency signal level problem at the downstream RF signal level is within specifications and you must spellcheck other key parts of the physical layer for analog channel problems this week looking for CTB composite Drupal beat or CSO composite second-order or other interference issues the television display itself can indicate such problems with analog services the description of the impairments later this module for digital channels means examining murder modulation error rate per bear bit error

rate or ratio quadrature amplitude modulation consolations ingress and someone to do this you'll need it HD essay or signal level meter with integrated cable modem your cable operator or a supervisor will instruct you on on to what murder Amber levels are acceptable but for 2 B6 qualm a minimum or level of 33 DB is often specified or greater than 33 DB is also even better while a pre-FEC FEC forward error correction berm a and B nonzero example 189 0E0 is perfect for Burr the post Debbie Sieber should be 04 downstream signaling and digital performance measurements are fine and in particular if the issue of the beauty only then you should suspect upstream communications and HD essay or SOM with integrated cable won't have the ability to communicate with the head and Jessica cable modem waterboarding MVA which media can be used to determine the device location the customer's premises is able to successfully communicate on the upstream to the head or and and or hub you cannot and you can and you can have many isolate the problem to the upstream indicators between CPE and had an MIC is a divide and conquer method the next section determine location of the upstream indications problem cable operator have adopted the whole whole home certification approach using the HD essay or SOM the above instructions can be simplified disconnect the DTA setup often collects cable use your D HD essay or SOM certify the connection passable test for triple play at whole home DVR services or mocha services if not then proceed to the line conquer method of the next section determine the location of the problem if the collection CPE passes all home a whole whole home certification test on the HD essay in of the lamb or alternatively if all the previous described test revealed no problems with the physical layer collects network that you've isolated problem to one or more of the following a single set-top box or DTA the customers display device television monitor or any other equipment between set-top boxes TV such as a VCR DVD player Blu-ray player I stereo equipment home theater and so any cables between any of these vices the 1st step is determine set-top box or DTA functioning properly by connecting a test TV or a HD essay or SOM support that functionality as well and we will connect this test TV connected to the output of the set-top box or DTA determine set-top box or DTA is functioning properly if the set-top box supports HDTV the customer TV is in a CTB then clearly the testing fee must also be a CTB ready or capable of for this approach to work alternatively you can connect the customer CV directly to the wall outlet for collects cable and bypass set-top boxes DTA to see the TV works properly with this approach only only works for analog or digital channels are not encrypted if no video was seen through the set-top box or DTA and analog and or unencrypted child can be viewed using the customer CV simply indicates a problem set-top box or DTA and possibly also the cables between the TV and the set-top box or DTA but since so much video content these days is encrypted is not conclusive that the customer CV does not work on all channels when connected directly to the wall at work outlet or collects cable line if you do have clear indication of a set-top box or DTA problem it may be due to a hardware or firmware update or configuration settings or maybe some static electricity wiped out the latter is easily checked by disconnecting set-top box or DTA from AC power cord with about 20 seconds plug the power cord back in his if the unit functions properly afterward is a set-top box or DTA appears to be function properly you've isolated the problem to the customers own equipment and cables between them or else a configuration between the set-top box in the customer TV this means it could be any of the following cabling or A/V connections presence of AC electrical power to the equipment configuration of the equipment customer own equipment self receivers TV CCTV 3-D TVs Blu-ray players DVD players VCRs A.B. switches video switchers Slingbox type devices game consoles note the same issue would not be resolved for example

with a PC or phone set when troubleshooting high-speed data or phone services customer account issues are also of a concern correct coding to the to the count for the services of the customer ordered common issues with customers own equipment customer owned equipment is no power incorrect A/V source selection A/V connections wrong poor automated connection on the wrong device batteries A/V connections missing incorrect A/V wiring use maybe we use component in the place of composite or vice versa or what may we took the composite mix up the colors incorrect equipment settings or programming example in isolation for customer owned video equipment when a device is a Blu-ray player or conventional BB players present be certain the TV is set to the appropriate input such as an HDMI or component VCRs connected in series make sure the TV and VCR to the correct set-top box output channel which is usually a channel 3 or 4 the problem turns out to be customer CV explained the issue is not the RF signal and show the operational test TV or radio operation on the H DSA or signal level meter if necessary you may compare the video display the customer TV against the test TV by utilizing an RF signal sorter understand that this adds an additional 3.5 DB a minimum loss or attenuation to the signal and it was mortal before adding the splitter it could be below threshold afterwards this watches no splitters if the RF signal levels are not correct for either the upstream and downstream and or the murder Amber is not consistent or indeed if it is prescribed signal performance metric failed in the above analysis or isolation then the next step is to locate and point at which locate the point in which the same level of performance becomes becomes out at specified levels this is a divide and conquer method of cable troubleshooting basically means that you can pick an intermediate location and check the signal level and or performance metrics there if okay the move down towards the CPE if not then you move upstream towards the top are various locations without the CPE it was a physical layer can have a fall as shown here since many times the problem is the customer's premises wiring which as we can see here for the diagram can have a wide variety of layouts and components the most logical place to start dividing the possible cases of fall is at the ground block highlighted here this will in effect divide the possible fault locations between those on the cable operators plant and those in the customer premises logic of the divide conquer method is simple composer is 10 possible Fairpoint between the top and the outlet for which the initial signal testing showed problems if you start with .1 the methodically went to point to them .3 and so on and it turned out that the problem was at .9 it would take you 9 measurements to get to the point when the other hand if you began with .5 founders downstream of of at that point then went to .8 and again found there was downstream and then your next measurement was .9 and avoid taking the 3 measurements to locate the problem thoughts on average the divide conquer method provides a faster resolution to the customer's issue during the divide conquer method be sure to look for uninspected variances and RF levels and/or more historically most of the troubleshooting calls will be spent troubleshooting RF signal level number for example murder would be 33 DB at all points tested the location of the fall to identify with them are going 22 and all points downstream of that location on the other hand are signal level and murder are not within's best gauges were measured at the ground block that usually means that the drop is faulty or the top is faulty measurement of the tab itself will quickly determine the top is okay the night but the drop is not note that if the problem is indeed due to the drop line would have your 1st clue would have been that all TVs in the home were experiences a problem with the customer had informed you of this during your initial analysis the above divide conquer examples using RF signal level and murder refers to the potential downstream problems if on the other hand the

issue is and upstream promising by the H DSA or signal level meter with integrated cable modem you could use a method connect the H DSA or signal of a medium to drop and going to the ground block and verify me upstream medications as shown here on the diagram if not the problem is either drop line of the top as shown here the diagram if yesterday injected out of the ground block and the problem is in the customer premises wiring should move inside your back to the divide and conquering an apartment work while he is also possible the problem isn't present while your present this is referred to an intermittent problem why you may not see the problem while on site you may still be able to locate the source of the intermittent problem by performing the complete suite of tests prescribed by the cable operator supervisor at the at all possible but appointed confirming for example that the signal attenuation from the top to the CPE is correct you can measure or calculate a lawsuit drop splitters and premises cables the murder Amber are within company specifications at the top and remain within specifications all the way to any and all CP devices in the premises I checked of the return spectrum at the top confirms that the card annoys and signal lawyers are within company specifications see the sections below on troubleshooting to return for more information confirm the fittings tagged and cables are installed properly before leaving the tab the drop has been properly installed in addition you also check the following F connectors compression fittings and look for removal of crimped fittings or hex fittings collectible jumpers check out wall plates in home amplifiers and passives loose connections poor or bad bonding of the ground block or bonding block it is time now to resolve the problem that was isolated during the divide and conquer assume for this example that you're already checked RF signal levels murder Amber and other levels at the top and have ruled out the view to the accident network has any issues several different examples of problem resolution are presented in the coming sides of the process can be seen in applied to other situations in example 1 in this example problem with the customer has a 2 way of acquiring a home that is failed in such a manner that the upstream is blocked while downstream is amplified normally by the method of the body conquering you identify the afar was at the point of failure and that while the downstream levels of performance metrics within company specifications at this point the upstream test showed that the upstream is not being transmitted through this device as many cable operators are moving to completely passive architectures for the home networks wherever possible your supervisor may direct you to rewire the home to eliminate multiple splitters in the premises and improve the signal levels without the need for replication if implication in the premises is still necessary based on the attenuation calculations you may still be directed to rewire the premises and use a company supplied amplifier at a point specified by your company or you may be directed to merely replace the faulty Apple fire example to suppose suppose instead of the problem has been located in single Pisa collect in the customer premises between the outlet in the home may home splitter and the Pisa coag is an older type shown here say series 59 cable company policy debate may be to replace a slide with newer series 6 collects and in fact may be to rewire the customer's premises is as required by your company specifications on the other hand you may see a connector that is clearly damaged or improperly connected characterize the 1st resolution tactic is to reconnect her eyes at that point and then check to see if signal levels and performance metrics are now within specifications at the Pisa coag is already a series 6 and will contact you can see it the connectors appear okay you have you may have replace this rotten which could be time-consuming was behind a wall even of the connector looks to be sound you may reconnect drive the cable on the 1st one event on the other end Becky signal levels at the periphery connectors agency of the

resolve the issue is putting new connectors on both ends and not resolve the issue than the collective lightly damage somewhere and must be replaced example for in this example assume that the set-top box was that a part of the source of the problem it turns out the set-top box is not properly provision for the services the customers ordered as a result they are not able to view a subset of channels the customer told you which channels they could not receive any check the specified channel RF signal levels and performance metrics using the channel map provided to you by the cable operator you verify that all signal levels and performance metrics are within specifications you note that the channels that can be received okay include analog digital set of standard definition and digital HD channels however then you know that all channels that can be received okay are in the basic tier of channels in the customer's point that it informs you that he's paying for a higher tier of channels simply call the office to check provisioning and verify the account reflects this in this case it does so you suspect the device without properly provision your request a reprovisioning of the set-top box and that solves the problem note that it may be tempting to show the customer the action is being taken by swapping out set-top box or cable modem the failure rate of such equipment is low so do not swapout such equipment as the 1st solution to problems often of the set-top box has been identified as the culprit it can be a configuration issue not a hearted hardware issue is really set-top boxes true because of the problem to cause a problem and that is not a configuration of provisioning issue before replacing it procedures and practices may vary from operator operator however so please work with the local management for guidance if there's any questions as to the method of problem resolution and repair in the final step to confirm the resolution and repair be sure to check all collects cable outlets to be sure their improper operation as a customer check services that were impacted prior to your arrival as well as other cable services such as BOD there are many on-demand movies that are free to customers to use a check BOD operation if on the other hand the beauty problems were with ordering a non-free movie encourage the customer to order a movie to Tessa set-top box ensure that it was not a problem in the billing of the customer's account any charge can be credited to customers account per your company's policy and the customer satisfied if the customer to sign off on the work order educate the customer on the corrective action that was taken resource services returned any faulty or other customer owned devices or cables that you replaced and explain why they were replaced be certain to ask customer if there's anything else you can do to make their cable experience better